IMT Institute for Advanced Studies, Lucca

Lucca, Italy

Multimedia Quality improvements for Next Generation Networks

PhD Program in Computer Science and Engineering

XXV Cycle

By

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2013

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IMT Institute for Advanced Studies, Lucca

2013

Dedicated to my lovely husband *Waqas* and our son *Sarim*.

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List of Abbreviations

2D	Two-dimensional
3D	Three-dimensional
3G	Third Generation Networks
3GPP	3rd Generation Partnership Project
4G	Fourth Generation Networks
AFDPS	Adaptive Fair Delay Prioritized Scheduling
ALM	Application Layer Multicast
AMBR	Available Maximum Bit Rate
AMC	Adaptive Modulation and Coding
AP	Access Point
API	Application Programming Interface
ARP	Allocation/Retention Priority
ARQ	Automatic Repeat-reQuest
ATM	Asynchronous Transfer Mode
AVC	Advance Video Coding
BER	Bit Error Rate

BGP	Border Gateway Protocol
BL	Base Layer
BWA	Broadband Wireless Access
CAC	Call Admission Control
DRR	Deficit Round Robin
DS-CDMA	Direct-Sequence Code Division Multiple Access
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
EL	Enhancement Layers
eNodeB	Evolved Node B
EPC	Evolved Packet Core
EPS	Evolved Packet System
FEC	Forward Error Correction
FIFO	First-In First-Out
GBR	Guaranteed Bit Rate
GPS	Generalized Processor Sharing
HeNB	Home eNode B
HOL	Head Of Line
IP	Internet Protocol
LTE	Long Term Evolution
MAC	Medium Access Control
MBR	Maximum Bit Rate
MCS	Modulation and Coding Scheme

MOS	Mean Opinion Score
NGN	Next Generation Network
NHAG	Network-aware Hierarchical Aggregation Graph
NS-2/3	Network Simulator-2 or 3
NXLO	Non-cross-layer optimization
OFDMA	Orthogonal Frequency Division Multiple Access
OSI	Open Systems Interconnection
P2P	Peer-to-peer
PDU	Packet Data Unit
PHY	Physical Layer of OSI model
PSNR	Peak Signal-to-Noise Ratio
QAM	Quadrature Amplitude Modulation
QCI	QoS Class Identifier
QoE	Quality of Experience
QoS	Quality of Service
QPSK	Quadrature phase-shift keying
RB	Resource Block
RED	Random Early Detection
RLC	Radio Link Control layer
RRC	Radio Resource Control
RRM	Radio Resource Management
RTCP	Real-time Transmission Control Protocol

RTP	Real-time Transport Protocol
SAE	System Architecture Evolution
SC-FDMA	Single-Carrier Frequency-Division Multiple Access
SDU	Service Data Unit
SIC	Successive Interference Cancellation
SIP	Session Initiation Protocol
SNR	Signal to Noise Ratio
SVC	Scalable Video Coding
TCP	Transport Control Protocol
TFRC	TCP-Friendly Rate Control
TTI	Transmit Time Interval
TXOP	Transmit opportunity
UDP	User Datagram Protocol
UE	User Equipment
UEP	Unequal Error Protection
UMTS	Universal Mobile Telecommunications System
VoD	Video on Demand
VoIP	Voice over IP
VPAL	Video Packet Adaptation Layer
WiViOpt	Wireless Video Optimality
WLAN	Wireless Local Area Network
WPAN	Wireless Personal Area Network

WPAN	Wireless Personal Area Network
WSN	Wireless Sensor Network
XLO	Cross-layer optimization

Acknowledgements

Firstly and above all, I would like to thank God for giving me health and all the internal resources needed for the fulfillment of this doctoral endeavor. All my gratitude to my beloved husband Waqas, who has always supported and encouraged me. Due to his love and patience, this dream has come true. Many thanks to my family, specially my brother Sagheer Ahmed, who raised me and helped me to form the human being I am now.

I am really thankful to my advisor, *Dr. Leonardo Badia*, for receiving me with open arms in Lucca, Italy, and for his commitment and generosity in sharing his wisdom. He has been an inspiration in both personal and professional aspects.

My acknowledgments to all the professors and employees of the Department of Information Engineering, specially Prof. Michele Zorzi and Daniele Munaretto, for the rich technical discussions and friendship.

I am very grateful for the financial support of the Italian government by means of the scholarship. Many thanks to Dr. Nicola Baldo and Marco Miozzo for their valuable guidance during my research period in CTTC, Barcelona, Spain. I am also specially grateful for the guidance of Dr. Andrea Zanella and Dr. Luca Sanguinetti, who have closely followed the progress of my doctoral work.

I would like to acknowledge the work of the anonymous reviewers of this thesis and the papers submitted to different conferences. Their valuable comments helped me to improve the quality of this work.

Last, but not least, many thanks to the friends I have known here in Lucca and Padova, specially my colleagues at the SIGNET Lab in University of Padova, for their help and friendship.

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Publications

1. Iffat Ahmed, Leonardo Badia, ”Analysis of Management Policies for Multicast Transmission of Scalable Video Content in Next Generation Networks”, in *Proceedings of The 9th IEEE International Wireless Communications and Mobile Computing Conference*, IEEE IWCMC 2013
   Abstract— Users of video multicast groups are highly heterogeneous in terms of individual channel conditions and requirements for video transmission. Their experienced quality may vary, making it a challenging task for the network to optimally configure the resource management. In this paper, we consider mathematical model to represent layered video content delivery in a multicast group. We compare various network policies to choose the optimum number of transmit opportunities and we investigate the role of feedback, which, if present, dynamically tunes the resource management. We analyze the actual perceived quality of the users as well as how their satisfaction levels vary in the multicast session. Simulation results show that the presence of feedback generally enhances the overall users quality; however, this improvement is heavily related to the resource allocation policy of the operator.

2. Iffat Ahmed, Leonardo Badia, Daniele Munaretto and Michele Zorzi, ”Analysis of PHY/ Application Cross-layer Optimization for Scalable Video Transmission in Cellular Networks”, in *Proceedings of The 14th IEEE International Symposium on a World of Wireless, Mobile and Multimedia Networks*, IEEE WoWMoM 2013
   Abstract— We investigate the optimization of video transmissions over cellular networks by using the H.264 Scalable Video Coding (SVC) at the application layer and an Adaptive Modulation and Coding (AMC) scheme at the physical layer. We analyze how the cross-layer optimization (XLO) of these two techniques together performs compared to a sequential and independent selection of video packets and Modulation and Coding Schemes (MCS) with no cross-layer optimization (NXLO), in terms of goodput and packet delivery delay. We formulate an analytical model based on a Markov chain representing the wireless channel, where each state is associated to a different channel quality corresponding to a set of possible choices of video layer and MCS. Our numerical results show that XLO significantly outperforms NXLO for video transmissions, thereby pointing out the strong need for cross-layer solutions in video transmission.

3. Iffat Ahmed, Cross-layer Optimization for Scalable Video Transmission in Next Generation Cellular Networks", in *Student Poster Session*, IEEE INFOCOM 2013
   Abstract— This paper tries to investigate that whether PHY/ Application cross-layer optimization really required for video transmission over next generation wireless networks? Or would a sequential allocation where optimization is independently performed at the PHY and Application layers work similarly? How does the cross-layer and non-cross-layer optimization perform also compared to the theoretical best allocation that one could apply, if the channel states and the user quality requirements were all known in advance? Is there a way to adapt to the channel variability? How do the unicast scenario extend to the multicast case? Given that a compromise in allocation must be found between the needs of all the users in the multicast group, it may be that cross-layer optimization is insufficient. Our numerical results show that XLO significantly outperforms NXLO for both unicast and multicast video transmissions, thereby pointing out the strong need for cross-layer solutions in video transmission.

4. Iffat Ahmed, Leonardo Badia, "A Quantitative Analysis of the State of the Art in Quality Provisioning for Multimedia Services over Next Generation Networks", in *Proceedings of the 9th ACM International Workshop on Mobility Management & Wireless Access*, ACM Mobiwac 2011
   Abstract— In this paper, we performed Meta Analysis of leading journals, conferences, letters and magazines to dig out the exhausted and least investigated areas within the umbrella of Multimedia QoS provisioning. We have tried to aggregate the result trends for Multimedia QoS according to a subject taxonomy, as well as several other relevant classifications (scenario, approach followed, and so on). The main motivation behind this work was to gather information and possibly identify the future research trends on this very important area.

5. Iffat Ahmed, Leonardo Badia, Nicola Baldo and Marco Miozzo, "Design of a Unied Multimedia-Aware Framework for Resource Allocation in LTE Femtocells", in *Proceedings of the 9th ACM International Workshop on Mobility Management & Wireless Access*, ACM Mobiwac 2011
   Abstract— In future 4G communication networks, it is expected that most of the bandwidth will be used to serve multimedia applications. Thus, there is a need for scheduling mechanisms, which can manage various types of multimedia communication (interactive, conversation, video streaming, etc.) and provide adequate Quality of Experience according to the application needs. The most promising candidate for 4G systems, i.e., Long Term Evolution, also integrates femtocells as a cost-effective solution for pervasive communication. In such a scenario, the implementation of effec-

tive scheduling mechanisms becomes even more important. Finally, before implementing scheduling policies on real devices, validation through simulation studies is often employed. For all these reasons, we investigated the implementation of an adaptive scheduling mechanism for an LTE scenario with femtocells within the well known ns3 simulator. This paper describes the preliminary steps of this activity.

6. Iffat Ahmed, Leonardo Badia and Khalid Hussain, "Evaluation of Deficit Round Robin Queue Discipline for Real-time Traffic Management in an RTP/RTCP Environment", in *ACM/IEEE European Modeling Symposium*, EMS 2010
   Abstract— Multimedia real-time traffic is deemed to be dominant in future communication systems. One of the reference applications to support real-time traffic is the Real-time Transport Protocol (RTP), which can be used to transmit multimedia contents on real-time basis. At the same time, Real-time Transmission Control Protocol (RTCP) is used for receiving feedback and getting information about the network. This paper proposes and evaluates a traffic management implementation in such an RTP/RTCP environment for congestion control. Deficit Round Robin queue discipline is used as the traffic management strategy instead of Random Early Detection and DropTail queue disciplines. A simulation campaign was performed to analyze the effects of implemented traffic strategies in RTP/RTCP environment and compare it with previous solutions. The obtained results highlight a significant difference in terms of jitter delay and packet losses and improvement the bandwidth utilization for real-time flows. Thus, we are able to provide quantitative evidence of the importance of the queue discipline to efficiently manage multimedia content.

Other Publications

1. Iffat Ahmed, Khalid Hussain, Nyla Khadam and Faraz Ahsan, Efficient Multimedia Traffic Engineering in Next Generation Networks, in: *International Conference on Information and Resource Management*, Conf-IRM 2009
2. Iffat Ahmed and Mansoor Alam Ansari, Load Balancing in Enhanced AMC to integrate Heterogeneous Wireless Networks in: *Springer: Wireless Networks, Information Processing and Systems*, Book Chapter, Vol. (20), PP. 110–117, 2008
3. Iffat Ahmed and Mansoor Alam Ansari The Generic Frameworks for Multimedia QoS in Wireless Networks in *Poster in Center of Excellence in Wireless and Information Technology Conference*, CEWIT 2008

4. Iffat Ahmed and Mansoor Alam Ansari, Provisioning Multimedia QoS in Wireless NetworksThe Meta Analysis in: *2nd International Conference on Information Communication Technology*, ICICT 2007
5. Iffat Ahmed and Mansoor Alam Ansari, Enhanced-AMC to integrate Heterogeneous Wireless Networks in: *2nd International Conference on Information Communication Technology*, ICICT 2007

Presentations

1. Iffat Ahmed, Cross-layer Optimization for Scalable Video Transmission in Next Generation Cellular Networks", in *Student Poster Session*, IEEE INFOCOM Apr 2013
2. "Wireless Video Optimality and Quality of Experience (WiviOpt)", in *Simula Research Lab, University of Oslo, Norway*, Apr 2013
3. "Cross Layer optimization of Scalable video in cellular networks", in *SIGNET Lab, Dept. of Information Engineering, University of Padova, Italy*, Mar 2013
4. "PHY/Application Cross-layer Optimization of Video Distribution over LTE Networks?, in *10th Italian Networking Workshop, Bormio, Italy*, Jan 2013
5. "The Meta Analysis of Multimedia Quality of Service (QoS) Provisioning, in *CTTC: Centre Tecnolgic de Telecomunicacions de Catalunya, Barcelona, Spain*, Jan 2011
6. "Evaluation of Deficit Round Robin Queue for Real time traffic and my research activities", in *SIGNET Lab, Department of Information Engineering, University of Padova, Italy*, Oct 2010
7. "Evaluation of Deficit Round Robin Queue Discipline for Real-time Traffic Management in an RTP/RTCP Environment", in: *ACM/IEEE EMS, Pisa, Italy*, Oct 2010

Abstract

Video is foreseen to be dominant in the Internet and Next Generation Networks, due to the increased usage of multimedia applications. The current Internet, and in particular the mobile Internet, was not designed with video requirements in mind and as a consequence, its architecture is very inefficient when handling video traffic. Not only is a policy optimization required, but it is also important to perform such an optimization in the proper manner. Therefore, providing Quality of Experience for such networks is an open issue and hot research area nowadays.

Our goal is to investigate the performance of the PHY/ Application cross-layer optimization, for which we developed an analytical model to optimize the number of timeslots needed for a video to be correctly decoded with enhanced quality. The wireless channel is modeled by means of Markov chain, whose state represent different channel qualities. We exploit Cross-layer (PHY/ Application) solution with respect to application layer information about scalable video layers, and taking user channel status for adapting channel rates. This problem gets more crucial when the case of multicast is considered, as the base station needs to harmonize the heterogeneous requirements of all the users and adapt transmission accordingly.

Performance is evaluated for various scenarios to investigate, what is the optimum number of time slots needed for the base layer of SVC, how does the feedback impacts on the end user perceived quality and user satisfaction level, and to what extend is Cross-layer optimization beneficial. Further, we evaluated how the unicast extends to multicast and its impact on end-user goodput, packet delivery delay and quality.

Chapter 1

Introduction

Multimedia services, such as video conference, video on demand, live streaming, are expected to be widespread which brings into play the provisioning of Quality of Service (QoS) or Quality of Experience (QoE) as a great challenge in future networks. Thanks to the evolution of broadband communication systems, such high demanding contents can be transmitted and viewed with acceptable quality. An important issue for future communication systems is the provisioning of QoS/QoE of multimedia applications, which are sensitive to these quality requirements. Particularly, video packets have stringent deadline requirements and are much prone to packet losses, therefore QoS/QoE for multimedia and particularly for video contents is a challenging and hot research topic in this era.

In this work, we address the interface between video services and the underlying network mechanisms that allow video services to efficiently customize the network behavior, thereby improving the user experience. Further, enhanced wireless access is studied to improve the video performance by exploiting the features of existing wireless technologies in coordination with video services.

The main goal of our research is to design solutions within novel mobile Internet architecture, which are aimed at efficient video traffic support. As video is expected to represent the majority of the traffic in the near future, the requirements of this traffic type must be directly accounted for, and specific enhancements for video should be introduced at all layers of the protocol stack, where needed.

We have done meta analysis research as a base study to find out research trends for Multimedia QoS and we found that in the Long-Term Evolution (LTE) network, it is still least investigated area. The possible reason could be novelty of the LTE networks or still not much work has been published in the selected leading journals/conferences could be another reason.

Our motivation is to provide Multimedia QoS/QoE in LTE networks which is future communication network through radio resource management along with Packet Scheduling. On the other hand, Cross-layer optimization is a broad and important subject, and is definitely a key issue in the LTE technology regarding the support of video services, which have inflexible constraints and QoS requirements. With our research we want to incorporate both the theoretical and practical aspects of multimedia contents delivery and we have developed our own simulation modules for this purpose.

Video users in multicast group are highly heterogeneous in terms of individual channel conditions and requirements for video transmission, making it a challenging task for the network to optimally configure the resource management. We consider a mathematical model for the choice of the optimum number of transmit opportunities, by using the Scalable Video Coding (SVC) at the application layer and an Adaptive Modulation and Coding (AMC) scheme at the physical layer,for both unicast and multicast environments. Based on a Markov chain representation of the wireless channel, each channel state is associated with a different quality corresponding to a choice of video layer and Modulation and Coding Scheme (MCS).

In case of multicast transmissions, the selection of the video layer is based on the aggregate channel conditions of all users and some coordination rules. We evaluate how the cross-layer optimization (XLO) of these two techniques together performs compared to a sequential and independent selection of video layer and modulation and coding schemes with non-cross-layer optimization (NXLO), in terms of goodput, packet delivery delay and quality.

We also define a taxonomy of user's quality perception and investigated that how users' satisfaction levels vary in the multicast session as well as the impact of feedback, which, if present, dynamically tunes the resource management.

1.1 Thesis Objective and Research Methodology

Multimedia QoS/QoE is a general field, and includes many sub fields, like Call Admission Control, Mobility Management, Resource Allocation, packet scheduling, flow management and many more. In the next generation networks, almost 90% of the contents would be Multimedia (or particularly video) over communication networks. Well known documents like the Cisco report [3] offer evidence of such a "video explosion" by impressively foreseeing that the amount of video content crossing global IP networks each month in year 2016 will amount to the equivalent of over 6 million years of video duration. Not only is a policy optimization necessarily required, but it is also important to perform such an optimization in the proper manner. Therefore, providing QoE for such networks is an open issue and hot research area nowadays.

1.1.1 Proposal

Keeping in view the state-of-the-art and our meta-analysis research, it is observed that multimedia, particularly video transmission quality improvement is a major challenge for the future Internet. The current Internet, and in particular the mobile Internet, was not designed with video requirements in mind and, as a consequence, its architecture is very inefficient when handling video traffic. It is the vision that, as video is going to represent the majority of the traffic, the future Internet architecture should be tailored to efficiently support the requirements of this traffic type. Specific enhancements for video should be introduced at all layers of the protocol stack where needed, ideally supporting an incremental deployment.

The Next Generation Networks are expected to exhibit multimedia contents by 90% as compared to best effort traffic, therefore Multimedia QoS provisioning is a hot research area nowadays in the networking research community. Based on the extensive meta-analysis, various subject taxonomies have been identified to dig out that how various researchers are providing multimedia QoS. By this study, the subject taxonomies have been defined focusing on Multimedia QoS provisioning and a generic framework has been defined for the purpose. From the quantitative analysis, it can be clearly depicted that packet scheduling and resource allocation have already been extensively investigated fields for multimedia QoS. However, network management, rate control, power control and multicast/broadcast management are still open research areas which are less investigated until now with respect to Multimedia QoS provisioning or QoE of end users. As far as research methodologies and models are concerned, simulation is the most popular methodology in this field to verify the results, whereas, mathematical, temporal and multi-tier influence diagrams are widely used models in this field. The selection of research methodology and research model is purely based on the topic of interest and subject scenario.

On the other hand, during the whole study regarding underlying network technology, we found that Multimedia QoS in the LTE network is least investigated area. The possible reason could be novelty of the LTE networks or still no work has been published in the selected leading journals/conferences.

One motivation is to provide Multimedia QoS in LTE networks which is future network. Cross-layer optimization is definitely a key issue in the long-term evolution (LTE) technology regarding the support of video services, which have stringent constraints and quality requirements.

As per our findings of the meta-analysis presented in Chapter 3, the least investigated network technology is LTE, which is due to the novelty of the underlying technology and needs to do more research.

Another motivation is to provide Multimedia QoS by investigating various subject areas, instead of focusing on only one subject area. That is, the work is done on combination of subject areas as defined in subject taxonomy. The efficient algorithms are introduced and extensive simulation campaign has been carried out to provide better QoS/QoE results for multimedia in Next Generation Networks (NGNs).

Fig. 1 represents the building blocks of Multimedia QoS. This framework is the result of meta-analysis in general (as discussed in Chapter 3), that multimedia QoS can be provided by all the areas defined in subject taxonomy.

1.1.2 Research Methodology

Initially, the least and extensively investigated research areas are figured out under the umbrella of multimedia quality provisioning. In the second phase, literature analysis is carried out in the selected subject taxonomies. Once the research frameworks/models are defined and analyzed, extensive simulation campaign is carried out to verify the results, according to our plan, we worked according to following steps:

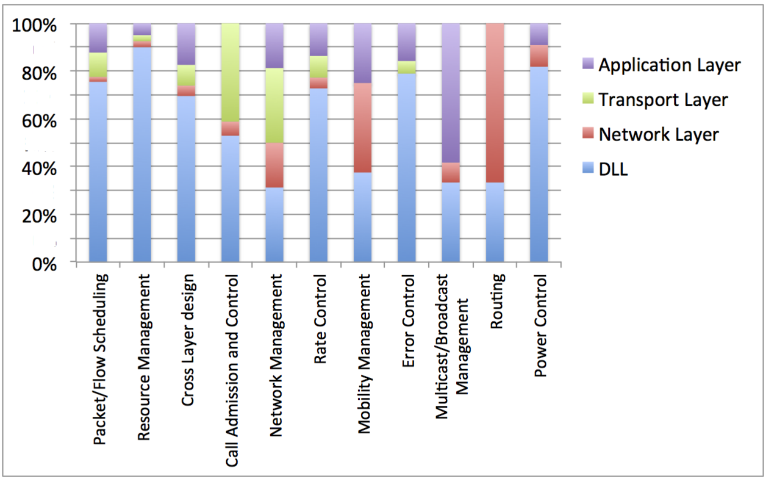


Figure 1: Building Blocks of Multimedia QoS w.r.t. OSI layers

i. Study the existing scientific research, already done for specific research taxonomy

ii. Definition of performance metrics for comparison with existing solutions. Represent the metrics to measure improvements introduced by our proposals with respect to existing ones. Therefore, a careful selection of Key Performance Indicators is done.

iii. Design the optimization framework for video QoE

iv. Build/prepare an appropriate analysis and simulation tool.

v. Design and execution of a significant set of simulations. In this last phase, we conducted simulations to validate results obtained analytically.

Given the incremental nature of commercial infrastructures, this research aims to perform improvements in the performance of Next Generation Networks, by exploiting various subfields/taxonomies under multimedia QoS, and that cumulatively will raise the network performance and Quality of Experience of video services to values unattainable with current systems.

In the near future video communications will be the predominant part of wireless mobile traffic and this fact is supported by a number of publicly available studies. Current infrastructures are not designed to deal with this traffic increase. The current Internet, and in particular the mobile Internet, was not designed with video requirements in mind and, as a consequence, its architecture is very inefficient for handling video traffic. Especially when a large part of this traffic is associated with multimedia entertainment (lets say: students watching musical videos), most of the mobile infrastructure is used in a very inefficient way to provide a simple service, thereby saturating the whole network, and leading to quality levels that are not adequate to support widespread user acceptance.

The main goal our research is to evolve the mobile Internet architecture for efficient video traffic support. As video is expected to represent the majority of the traffic in the near future, the future architecture should efficiently support the requirements of this traffic type, and specific enhancements for video should be introduced at all layers of the protocol stack where needed.

1.2 Contributions of the thesis

Guided by the objectives, meaningful and novel contributions of the thesis are manyfold. To sum up, this proposed research study makes the following contributions

i. We model the network management policies based on the individual requirements of the users and defined taxonomy of users.

ii. We model the multimedia-aware resource allocation for LTE Femtocells.

iii. We investigate the effect of feedback, i.e., how it impacts on user perceived video quality, and any improvements brought to the QoE of users

iv. We performed Meta-Analysis of existing research work in Multimedia quality improvements. It provides future research directions for upcoming researchers.

v. In addition, we address the following questions:

   (a) Is PHY/Application cross-layer optimization really required for video transmission over next generation wireless networks? Or would a sequential allocation where optimization is independently performed at the PHY and Application layers work similarly?

   (b) How do cross-layer and non-cross-layer optimization perform also compared to the theoretical best allocation that one could apply, if the channel states and the user quality requirements were all known in advance? Is there a way to adapt to the channel variability?

(c) How do the unicast scenario extend to the multicast case? Given that a compromise in allocation must be found between the needs of all the users in the multicast group, it may be that cross-layer optimization is insufficient.

The design of the video services tries to provide the bridge between video applications and the core network mechanisms making use of enablers for the communication with the other modules. Our performance evaluation results show that XLO significantly outperforms NXLO for both unicast and multicast video transmissions, affirming the strong need for cross-layer solutions in video transmission. Further, the presence of feedback generally enhances the overall users quality; however, this improvement is heavily related to the resource allocation policy of the operator.

1.3 Research structure of the thesis

The structure of the thesis is discussed as follows:

Chapter 2 introduces the scalable video coding concept and cross layer design approaches

Chapter 3 discusses the meta-analysis of leading journals, conferences, letters and magazines to dig out the exhausted and least investigated areas within the umbrella of Multimedia QoS/QoE provisioning. The main motivation behind this work is to gather information and possibly identify the future research trends on this very important area.

Chapter 4 describes our multimedia-aware flow/packet scheduling mechanism in LTE Femtocells. On the other hand, it also discusses our evaluation of Deficit Round Robin (DRR) queueing discipline particularly for multimedia applications over Real-time Transport Protocol (RTP) and Real-time Transmission Control Protocol (RTCP).

Chapter 5 proposes the analytical model for optimizing the resources particularly with respect to Scalable Video Coding (H.264/SVC). It also describes the resource allocation mechanism, and various optimization policies, keeping in view the cross-layer and non-cross-layer designs

Chapter 6 discusses the analytical and extensive simulation results. Performance of the proposed framework is evaluated by means of simulation and various results regarding video quality, delay, goodput are presented. We also provide results for availability of feedback and its impact on video quality. Video quality in multicast transmission is also evaluated and compared to unicast.

Finally, **Chapter 7** concludes the findings and discusses further evolutions for future research.

Chapter 2

Scalable Video and Cross Layer Design

In the current Internet structure, all the contents are treated equally, hence, there is no support to provide efficient video delivery and enhance the Quality of Experience (QoE), i.e., the actual quality perceived by the users, directly related to the user satisfaction [4]. Due to the proliferation of enhanced wireless technologies and video codecs, the demands are shifting towards user centric approaches, where satisfaction of requirements is of much importance compared to traditional Quality of Service (QoS) improvements through a network centric approach.

Most of the research on video services usually considers point-to-point scenarios, and less investigation has been done for multicast scenarios. Similarly, many studies have been done to improve the quality of transmitted video by using mechanisms for efficient resource allocation [5], congestion control [6] or bandwidth/rate management [7]. Multicast communication is an important solution for distributed multimedia applications, particularly video, to efficiently use the network resources by exploiting spatial and content redundancy of the user requests.

Multimedia applications are sensitive in nature and highly demanding, therefore efficient resource allocation schemes are required with special treatment for being transmitted over communication networks. Not only the policy optimization is necessarily required, but it is also important to perform such an optimization in the proper manner. Depending on the specific scheme, for multimedia contents, and specially video, there may be a strong impact on the perceived quality by the end users. This imposes to consider models of packetized traffic where the data stream is not necessarily homogeneous, but instead each packet, depending on its content, has different roles and priorities.

2.1 Scalable Video Coding/H.264

Among the unique characteristics of video traffic, we focus on its layered structure, with particular reference to the H.264 Scalable Video Coding (SVC) [8], [9] which is an extension of the H.264 Advance Video Coding (AVC). Compare to other video coding standard, SVC provides significant reduction in the bit rate which is needed to present certain perceptual quality of video.

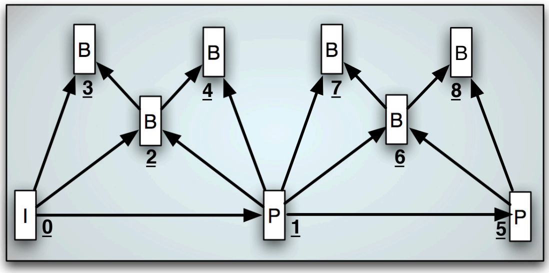


Figure 2: Hierarchical-B with GOP=4

Each video frame is divided into 8x8 pixels, known as blocks, such blocks are then grouped into 16x16 macroblocks. Macroblocks are grouped horizontally into slices which have similar average block levels. A frame is formed by combining multiple slices, and such kind of frames are known as "I" frames. On the other hand, "P" frames are predicted on prior knowledge about the I or P frames plus the additional data for changed macroblocks. "B" frames are bidirectionally predicted frames depending upon the position of past and future frames macroblocks. The Intra (I frame), Predictive (P frame) and Bi-directionally predictive (B frames) are the coding types. As an example in Fig. 2, let's consider the Hierarchical-B type coding structure, with the Group of Picture (GOP) size 4. The GOP size refers to the number of pictures that can cover all the level of prediction. The coding type (letter I,B or P) is mentioned inside the box.

As in the figure, the B frames need information from the past and future as well, therefore the future frame needs to be encoded before B frame. As depicted in the Fig. 2, there are three B frames requiring information from I and P frames, therefore more encoding delay will incur. Thus, the numbers with the boxes shows the sequence of encoding these frames. Hence, before encoding the second P frame, there will be a delay of encoding three B frames. This kind of problem is mitigated by SVC temporal scalability concept, which is illustrated in Fig. 3, in which stream is encoded in terms of temporal layers.

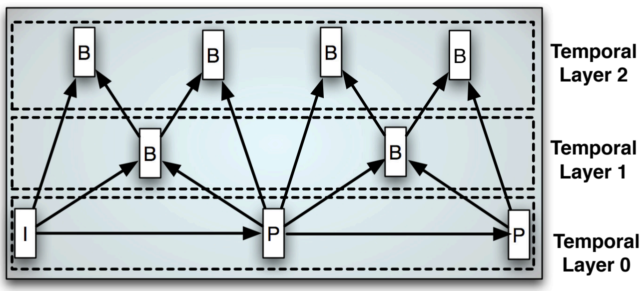


Figure 3: Temporal Layers in Hierarchical-B Coded Stream

In SVC, the bit stream is divided into layers, i.e., one Base Layer (BL) providing base video quality and multiple Enhancement Layers (ELs) bringing additional incremental quality. ELs are conversely dependent on the BL, since their decoding is useless if the BL is not fully decoded in the first place. As long as the users are receiving SVC base layer contents, they are able to decode the video, and when they receive the SVC enhancement layers, they actually start enhancing the quality of received video contents. The decoding of each EL frame necessarily relies on the decoding of the corresponding BL frame. Instead, upon losing an EL packet, the receiver will still be able to decode the video flow. Hence the quality decrease would be relatively minor; in the worst case, the basic video quality of playing only the BL is still guaranteed.

2.1.1 SVC Scalability Characteristics

The term "scalable" here refers to the phenomenon that, if a part of the video bit stream is removed but the whole stream can still be decoded, reconstruction at the receiver's end and reproduction are possible, albeit with lower quality.

SVC has three [10] major scalability characteristics: Temporal scalability involves the partitioning of video frames into layers, in which the base layer is coded with certain basic framerate and the enhancement layers are coded with respect to base layer temporal prediction. Secondly, having bit streams in a way that various subsets of such a bit stream represent different spatial resolution, as depicted in Fig. 4 is known as spatial scalability, whereas the property that a sub-stream can provide the same spatio-temporal resolution but with lower fidelity, that is, lower Signal to Noise Ratio (SNR), is known as Quality Scalability. [8]

SVC video can be adapted to heterogeneous kind of end users' devices [10], as depicted in Fig. 5, where, depending upon the capability of receiving device, it can decode the layers of video, and so receive the quality accordingly.

The literature is abundant with solutions addressing SVC quality improvements at the receiver's end, or rate adaptation in wireless transmission via Adaptive Modulation and Coding (AMC) schemes. However, most of these contributions target *either* of the topics (that is, SVC adaptation without considering modulation and coding scheme or vice versa). There are indeed some cross layer solutions for resource management in multimedia networks, however they mostly deal with improvements of

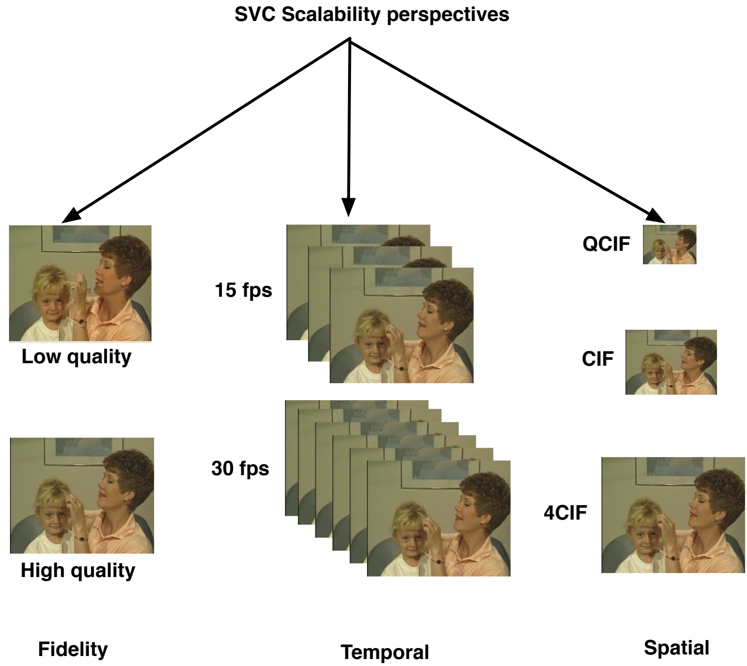


Figure 4: SVC Scalabilities

the overall QoS, without any consideration on the user perceived quality, i.e., the QoE. Some relevant references to the present work are discussed in this section.

To improve the performance of SVC video source, [11] provides a solution using a cross layer approach, which prioritizes the important portion of scalable video. It uses a fuzzy-based controller at the application layer for adapting the rate of the video content, based on the information gathered from the lower layers of the protocol stack. Simulation results are shown for sent/received bit rates and goodput. However, here the channel is an idealized pipeline, and the optimization is limited to the application layer, without any consideration on the wireless channel. Moreover, SVC is considered as an adaptive application layer, without

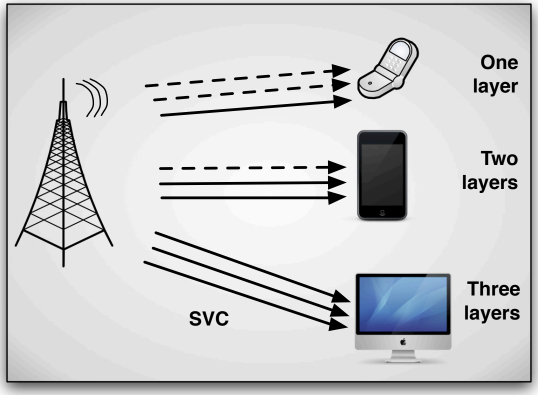


Figure 5: SVC Adaptation

any differentiation between BL or ELs.

Also, [8] investigates how enhancement layers of SVC bit stream can improve the perceived user QoE. This work shows that if the relationship between BL and ELs is properly accounted for in the resource allocation scheme, the resulting video quality is considerably increased compared to the use of the codecs H.264-AVC or MPEG-4; however, wireless channel models are not considered.

In [12], solutions are provided for video multicast in infrastructure based broadband wireless networks, where layered hybrid Forward Error Correction/Automatic Repeat reQuest (FEC/ARQ) is used to combat packet losses and different resources (and MCS) are assigned according to different video layers using Orthogonal Frequency Division Multiple Access (OFDMA) techniques. This approach is shown to improve the network management; the role of feedbagck about video quality is not considered.

A resource allocation scheme for scalable video has been presented by [13], which allot the modulation coding according to the video layer to be transmitted, providing an efficient allocation for the timeslots. The proposed solution is compared to the single layer video with fixed modulation and coding scheme.

State of the art of multimedia downlink scheduling is an extensively researched topic which involves packet scheduling/queueing management and/or prioritization schemes for particular technology like, Wireless LAN, Wimax or Universal Mobile Telecommunications System (UMTS). [14] provides cross layer solution by using SVC in Broadcast/multicast environment. Delay, throughput and fairness are considered as metrics for QoS provisioning. However, no evaluation on the actual perceived quality of the end user is performed. [15] utilizes multiple radio interfaces (including UMTS and WLAN) for SVC layers transmission, deciding which layer to transmit by considering the channel conditions of all the interfaces. The receiver then combines the SVC layers received from all of the radio interfaces, therefore involving the rate variation but not according to the BL and EL requirement, specifically. A Video Packet Adaptation Layer (VPAL) solution is proposed by [16] for multicast environment in WLANs, where the video rate depends on the channel transmission rate. However, this last parameter is purely based on feedback of previous transmission and not on the actual channel quality.

2.2 Adaptive Modulation and Coding

Future wireless networks will also exploit improved data rates offered by the ability to closely follow the channel dynamics. High-order modulations can improve the spectral efficiency of the transmission, however they also increase the bit error probability, which is undesirable for video contents given their specific error sensitivity. As noted before, the incremental structure of the content may cause error propagation in the following video frames, and therefore unacceptable quality fluctuations and artifacts at the end users. Thus, Adaptive Modulation and Coding (AMC) schemes are required to achieve high data rates. Due to the diverse channel conditions at the end users, we can adaptively select channel rates and select the layer of the SVC source accordingly.

Rate adaptation mechanisms connected to the underlying channel have been investigated for instance by [16] and [17], where it is exploited that rate variations during the transmission have a strong impact on the end user perceived quality [18].

The study of sudden large bandwidth fluctuations in Wireless LANs suggested experiments using SVC videos on a WLAN testbed as discussed in [19, 20], which claims that the base layer must be encoded with the highest possible coding rate and that applying simple priority queuing/scheduling (to prioritize the important packets) can improve the quality of the users. Similarly, such solutions do not consider a cross layer approach and do not assess the multicast performance.

To incorporate accurate video quality metrics, [21] provides a framework for rate allocation and also involves cross layer design and tries to provide differentiated QoE for various video sources. The emphasis is on maximizing a weighted QoE value for each video source considering Medium Access Control (MAC) layer scheduling. However, the paper does not discuss the impact of the number of transmit opportunities on QoE. All, these contributions do not exploit packet differentiation at the application layer, i.e., packets classified into BL and EL.

2.3 Long Term Evolution (LTE) — Next Generation Networks

4G systems are often identified with the Long Term Evolution (LTE) of 3G cellular systems, such as Universal Mobile Telecommunications System (UMTS). LTE offers multicarrier approach for multiple access. For the downlink, it uses Orthogonal Frequency-Division Multiple Access (OFDMA) and Single-Carrier Frequency-Division Multiple Access (SC-FDMA) in the uplink direction [22]. LTE incorporates the evolution of radio access with the help of Evolved-Utran (E-UTRAN) [23], therefore, LTE in conjunction with System Architecture Evolution (SAE which includes Evolved Packet Core (EPC)) is known as Evolved Packet System (EPS).

EPS uses the concept of bearer, which is basically the flow for traffic with defined Quality of Service (QoS) requirements for the User Equipment (UE). According to 3GPP standards, the QoS parameters for LTE systems are: QoS Class Identifier (QCI), Allocation/Retention Priority (ARP)), Guaranteed Bit Rate (GBR)), Maximum Bit Rate (MBR)) and Available Maximum Bit Rate (AMBR)). The QCI involves following parameters for each bearer type: Resource type, Priority, Packet Error Loss Rate and Packet Delay Budget.

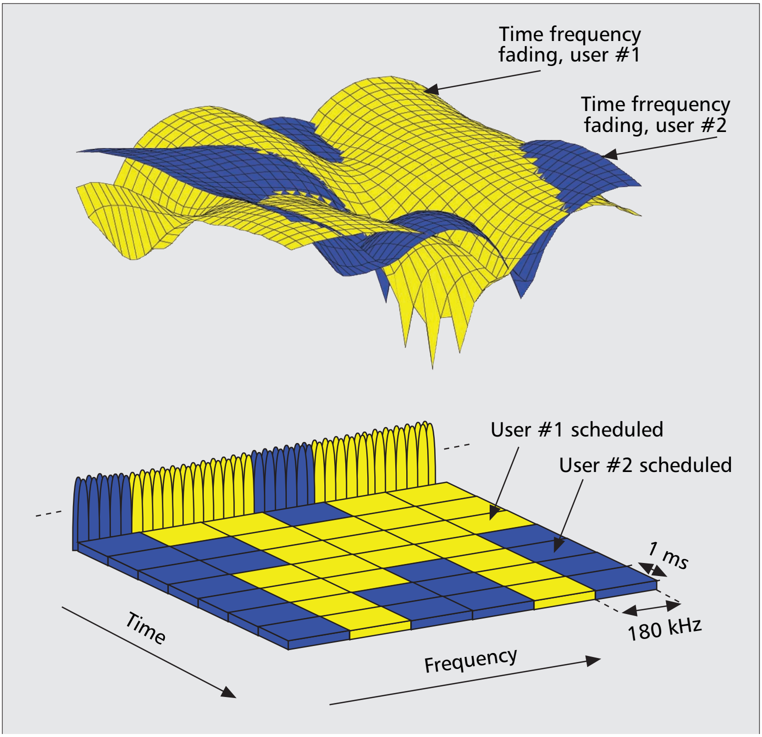


Figure 6: Channel Dependent LTE Resource Scheduling

Scheduling of radio bearers is done in eNodeB of E-UTRAN, so that resources can be allocated according to their QoS requirements and availability in eNodeB. The eNodeB is the intermediate point between all the users and core network; therefore, radio resource management takes place on eNodeB.

By nature, radio communications have fading, due to which quality of radio channel varies in time, space and frequency domains (as illustrated in Fig. 6, [24]. Since, LTE utilized Orthogonal Frequency-Division Multiple Access (OFDMA), therefore, LTE can exploit the channel depending scheduling by using time and frequency domains. For each sub-frame the LTE scheduler decides the users allowed to be allowed for transmission and schedules/assigns the Resource Blocks (RBs), depending on the scheduling algorithm. One of our work, related to flow/packet scheduling is presented in Chapter-4.

2.4 Cross Layer Design

Cross layer design can follow various approaches, as shown in Fig. 7, [1]. OSI layers can interact in different ways, some cross layer designs focus on merging the two adjucent layers or some approach to proposing the interfacing between the layers. [1] represent a survey of various Cross-Layer design and how the researches are done in the context of its design. Fig. 7(F) presents the PHY/ Application cross layer design, which has to span across all the layers of OSI protocol stack.

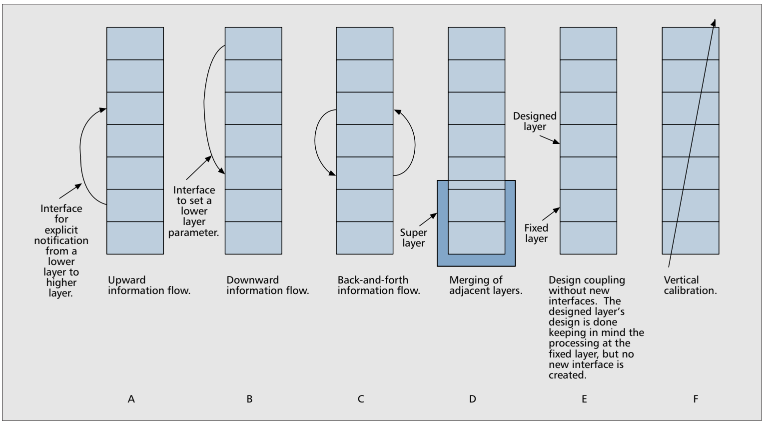


Figure 7: Kinds of Cross-Layer Design, taken from [1]

At a first glance, these problems are similar to general packet scheduling frameworks that have been proposed in the literature [25], [26]. However, while this kind of resource allocation has been deeply explored in the literature, it was often targeting Quality of Service (QoS) provisioning, which usually refers to improving goodput (i.e., the net fraction of throughput after removing coding overhead), link utilization and packet delay, but it does not necessarily improve the end user perceived quality. Instead, all the procedures that we perform to specifically address video content target the improvement of the Quality of Experience (QoE), i.e., how the end users perceive their received video content.

To understand the difference between a pure QoS and QoE enhancement, we consider the case where a good channel quality is exploited for transmitting several enhancement layer packets. While the goodput is improved, and therefore the QoS, the resulting quality may be totally unaffected if the base layer is not correctly received. This is why we advocate the need not only for a generic "cross-layer" optimization, but more specifically a PHY/ Application cross-layer optimization, which is even more challenging as it has to span through the entire protocol stack.

Taking advantage of adaptive channel rates and scalable video layers for quality enhancement can significantly improve the perceived quality experienced by the users. For this reason, we try to exploit both features to provide enhanced QoE for scalable video while taking into consideration varying channel conditions of the individual users.

2.5 Unicast and Multicast Environment

This problem gets more crucial when the multicast case is considered. Here, instead of looking into individual channel conditions and the packets of the available video stream for the single user, the assignment of base layer and enhancement layers must be coordinated among all the users in the multicast group. Thus, a suitable trade-off must be found among their channel conditions, and their diverse video quality requirements.

To improve the quality of multicast streaming for adaptive video, a cross-layer architectural solution is proposed in [27], which incorporates signaling between PHY, MAC and application layer and tries to improve delay, jitter and packet loss. At the same time, the amount of required overhead is investigated, seeking for a compromise on the signaling increase due to cross-layer information exchange.

Another similar approach is that of [28], which uses dynamic adaptive streaming with SVC to provide the cost effective solution. It focuses on improving the video segment quality according to the user download rate. The research variables considered are quality variation, performance, truncated normal rate, and so on, but there is no consideration on a wireless channel model nor multicast analysis, as the solution focuses on a single generic transmission between the client and the server.

Using a cross-layer feedback approach, [29] tries to estimate the user received video quality. The transmitted video signal and received video quality are monitored in three phases, i.e., signal encoding, transmission on lossy channel and the decoded signal. Such an estimation can be used to improve the performance of video transmission from the user perspective.

A utility (representing QoE) maximization problem is considered by [30], which focuses on user-dependent and session dependent utility functions. It tries to allocate resources efficiently to multicast groups and investigates the performance over average user rate, based on the number of groups. Even though it analyzes the multicast user performance, it does not investigate packet differentiation into BL and EL packets, as we do.

Feedback of video services in a multicast environment is also investigated by [31] and [32]. The former focuses on congestion control mechanism in Asynchronous Transfer Mode (ATM) networks, whereas, the latter approaches the layered video multicast problem through game theory and deals with error control.

Finally, [33] provides a framework to mitigate transmission impairments or avoiding error propagation at the receiver. The focus is experimental and the considered scenario is video conferencing over a Wireless Local Area Network (WLAN). However, this work does not exploit the layered structure of the video content, which induces some packet prioritization, nor it considers different user requirements within the same multicast group.

Thus, video quality improvement becomes more crucial when the video is transmitted on lossy wireless channels varying for all users, to heterogeneous devices, with heterogeneous video requirements in a multicast transmission. Therefore, pointing out a strong need for efficient resource management to have optimal network utilization and better QoE for video users.

Chapter 3

Background and Motivation

Due to high speed Internet and growing wireless multimedia communication systems, future communication systems are expected to transmit large amount of multimedia traffic such as video, voice and text with a variety of Quality of Service (QoS). An important issue for future communication systems is the provisioning of QoS of multimedia application which are sensitive in nature with respect to their quality requirements. Particularly, video packets have stringent deadline requirements and are much prone to packet losses, therefore QoS for multimedia and particularly for video contents is a challenging and hot research topic in this era.

Various studies have been carried out to provide multimedia QoS in next generation communication system [34], [35], [36] and [37]. The work presented by [38], [39] and [40], mainly focused on packet scheduling algorithms but also provide cross layer design to enhance the quality of service for real time contents. Whereas, [35] tries to improve the video services in the mobile embedded systems. It provides a flexible QoS assessment framework in wireless networks for the broadcast services of video under the cross-layer design approach.

The Fig. 8 below represents four key video services, namely Video

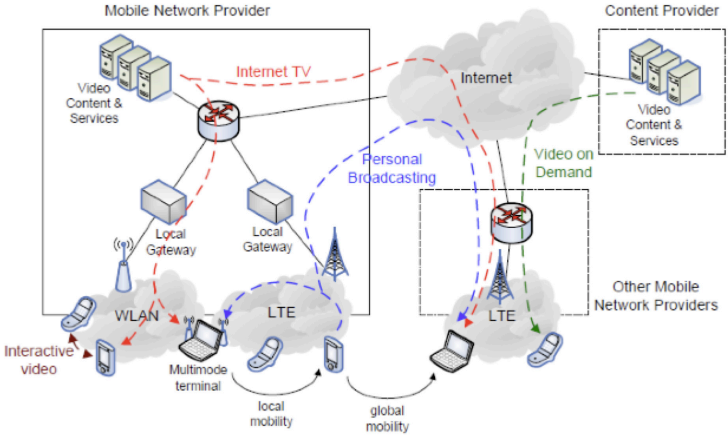


Figure 8: Video Content Delivery Network Overview — [2]

on Demand (VoD), Internet TV, Interactive Video and Personal Broadcasting. The associated technical problems that need to be solved for this vision to come true cover all the areas of mobile communications, from wireless access enhancements, efficient mobility management, and transport optimisation, to video distribution mechanisms.

Multimedia QoS is vast field and all the results are scattered, therefore, we are motivated to provide a generic framework and explore the subject taxonomies, of how different research works are being carried out in recent years for Multimedia QoS. In this regard, meta-analysis (defined below) is considered as a preliminary study to get the directions for future work and dig out the least exhaustive subject areas for Multimedia QoS.

3.1 Theoretical Framework

One of the objectives of the preliminary research is to develop the taxonomy of how various researchers are providing Quality of Service for multimedia contents, based on the literature review, and then conceptualizing the subject area, which are least explored. A total of 210[1] research articles have been reviewed/examined thoroughly and sorted out according to the usage of model, research methodology and research variables. The taxonomy of models and research methodologies is depicted from the [41]. Based on extensive examination of previous research work, following subject areas are categorized as the taxonomy of Multimedia QoS, in Table 1.

Table 1: Subject Area Taxonomy

S. No.	Taxonomy
1	Call Admission Control
2	Cross Layer design
3	Error Control
4	Mobility Management
5	Multicast/Broadcast Management
6	Network Management
7	Packet/Flow Scheduling
8	Power control
9	Rate Control
10	Resource Management
11	Routing

3.1.1 Taxonomy of Subject Areas

The subject area are described here briefly:

Call Admission Control

The Call Admission Control (CAC) is network management strategy, which is used to control or limit the number of call connection and to avoid the network congestion so that QoS can be improved, as presented by [36], [42], [43] and [44]. The new call admission control scheme proposed by [36] refers to the pre-computation of traffic scenario, online sim-

[1]Careful selection of such articles has been carried out from the leading conferences and journals for the period 2006-2013

ulation and decision making process about the call admission. It investigates on how to provide better QoS in terms of better bandwidth allocation and minimum standard deviation as well as it provides results for autocorrelation.

On the other hand, [45] mainly focuses on the traffic classification and then jointly performs packet scheduling and call admission control for uplink traffic in IEEE 802.16 networks. They choose the following performance metrics: latency, throughput, no of connections and the call blocking probability. Even though, it provides adequate results for QoS provisioning but more work can also be done regarding the communication and interaction between the scheduling algorithm and congestion control mechanism to improve the QoS as much as possible.

Cross Layer design

The term Cross-Layer design is well defined and justified by [1], [46], and [47] as *Protocol designed by the violation of a reference layered communication architecture is cross-layer design with respect to the particular layered architecture*. Here, "violation" means to create new interfaces between the Open Systems Interconnection (OSI) layers while possibly jointly tuning the system parameters across layers. Many of the recent researches done for multimedia QoS are relying on cross layer approach, because of the freedom to play at any layer and tune special parameters for real time and delay sensitive contents.

The good feature of Cross Layer design is the flexibility and autonomy. It also allows improvements in the optimal use of resources. On the other hand, the disadvantageous point of Cross Layer design is the violation of design and standards. It breaks layering structure of network, therefore, it can also simplify the design.

Error Control

In the packet communication networks, the packet transport service is not much reliable, therefore, QoS cannot be guaranteed specially for multimedia packets. One of the major techniques for error recovery is Forward Error Correction (FEC). [48] provides a model based on the analytical approach for the efficiency evaluation of FEC coding along with the combating packet losses. Similarly, [49] also studies the FEC block error, transmission rate and QoS, and provide results in terms of latency, block error probabilities, packet losses and estimation error.

Automatic Repeat-reQuest (ARQ) delay **??**, can play an important role in terms of QoS in terms of correlated arrivals of packets and error, particularly for real time contents. [50] presents the statistics for such a problem and provides mathematical framework work along with the numerical results based on queueing delay, arrival burst length, arrival rate, error burst length and delivery delay. It performs the analytical investigation of the packet delay statistics specifically for the Selective Repeat ARQ scheme with non-instantaneous feedback, particularly on the correlation of both channel errors and the packet arrival process.

Mobility Management

Mobile devices cannot rely on a single network, since by nature such devices are mobile, therefore they may make use of various networks and connectivity options while traveling. *Mobility management refers to enabling the user equipment for keeping the network connectivity when moving and changing its point of attachment* [51]. Various researches have been done on mobility management, which can be segregated as the layer on which these have been managed. For example, handoff through Session Initiation Protocol (SIP) is managed on application layer, Transport Control Protocol (TCP) migration is managed on Transport layer, whereas Mobile IP is managed on the network layer. [51] and [52] are focused on application layer mobility management, however [53] and [54] mobility management solutions are based on network layer primarily.

Due to wide variety of networks and heterogeneity of capabilities of each network, keeping an a-priori knowledge of changes can significantly improve the network usage and its scheduling. For example, a solution called "BreadCrums" has been proposed in [55] to observe the daily movements of the persons and learning their movement paths. They experimented this prototype, and showed improved results in terms of accuracy of learning the mobility patterns and less power consumption.

Multicast/Broadcast Management

Application Layer Multicast (ALM) schemes are becoming very interesting when the matter of streaming media is considered involving the transmission of multimedia streams to the large number of clients. The ALM mechanism involves the initial construction of a tree initially and then the transmission of the multimedia contents to the clients via the multicast tree. Network-aware Hierarchical Aggregation Graph (NHAG) is proposed by [56], to ensure the successful and efficient delivery of multimedia streams and reliability of the contents.

Due to the heavy traffic and wireless nature, the quality can be degraded, therefore [57] presents the concept of having multiple multicast trees to provide better QoS to the end user. This work mainly focuses on the reliability of the contents and throughput. Broadcasting mainly refers to the distribution of multimedia contents to a wide range of audience or receivers. There are various broadcasting schemes, for example, heuristic broadcasting or periodic broadcasting [58]. The work presented by [58] involves the user requirements about the multimedia contents and then switching between various broadcasting schemes.

Network Management

In the Next Generation Networks (NGNs) quality assurance is a major challenge due to media-rich contents and varying QoS requirements. Therefore, network management and tuning the network parameters to enhance the Quality of Experience (QoE) are important aspects. The QoE refers to quality of user perspective, whereas QoS refers to the mechanism employed for the management of network conditions and service differentiation [59]. According to [60], a network coding policy to achieve asymptotic capacity is devised, while maintaining the queuing and decoding delay. It focuses on provisioning of appropriate results in terms of packet delivery and expected throughput.

Future communication systems are expected to transit to wireless technologies; in wireless system, interference can play a major role to enhance or degrade the quality of contents. In this regard, [61] tries to decrease the interference and enhance the QoS for multimedia contents. It tries to analyze the quality by including the Successive Interference Cancellation (SIC) to the coherent and non-coherent modulations and creates the architecture for multi-code Direct-Sequence Code Division Multiple Access (DS-CDMA). This solution provides improvements for Bit Error Rate (BER) and Signal to Noise Ratio (SNR).

Packet/Flow Scheduling

Packet scheduling is an important aspect with respect to QoS provisioning. Usually, traffic differentiation (multimedia or best effort) on the Medium Access Control (MAC) Layer along with the queue characteristics is considered as point of research [62], [63] and [64]. However, [65] focuses on lower layer details along with the packet scheduling algorithm. It proposes an opportunistic scheduling, which takes advantage of physical channel condition, and so provides the QoS based on a queue and channel aware packet scheduling algorithm to maximize the channel utilization.

For the Internet applications, the streaming contents and file sharing/downloading are becoming more and more famous in the peer to peer networks. Content management in the P2P networks, therefore, play very important role for efficient delivery of multimedia and real time contents. Some of the challenges and architectural design issues have been discussed in [66] for peer to peer systems regarding content scheduling. The proposed solution not only measures the system performance but also takes care about the user satisfaction and Quality of Experience (QoE).

In the packet communication networks, the channel is stochastically shared by various media flows and these flows may have varying characteristics. Therefore, dynamic changes in throughput and delays may occur, and since multimedia traffic is sensitive to delays and throughput therefore network delays/dynamics may also effect the QoS for such media flows. In this context, the flow management is done by [67], [68], [69] and [70] to provide better QoS for multimedia flows.

Generalized Processor Sharing (GPS) is a work-conserving scheduling discipline in which multiple traffic classes share a deterministic server. GPS scheduling algorithm for traffic flows to enhance the multimedia QoS is designed by [71]. It considers server utilization, buffer overflow management, queue occupancy and delay violation probability as QoS performance metrics.

Power control

Power consumption is nowadays a critical issue for every electronic device; yet multimedia applications are resource hungry and are considered very power consuming. In wireless networks, the devices are mostly run on battery power. The power consumption of Access Point (AP) is usually higher, because AP has to be alive even when there is no activity in the network. Therefore, some power management can be done on AP. [72] tries to provide power efficient optimal solution for the AP as well as for station specially for real time contents (particularly for H.263 video) to enhance the QoS.

Similarly, [73] also provides solution for power management, while working on MAC layer in the Wireless Local Area Networks (WLANs). The proposed solution only focuses on the power management of the station, it does not cover the power management for the AP. It considers following performance metrics, data load, dropped ratio, frame delay and number of stations admitted.

Rate Control

Rate control mechanism are used for congestion avoidance and its control. Well known transport protocol TCP has also introduced congestion control mechanism which is TCP-Friendly Rate Control (TFRC). The research work [74] presents the extension of TFRC, named as MulTFRC. It uses steady state throughput equation to derive the throughput of TCP flows.

On the other hand, [75] present the dynamic multi-rate aspect for the multicast transmissions. It provides the solution for WLAN, for the dynamic rate adaption of multicast multimedia traffic. It allows real time contents to be transmitted at higher rate as compare to the best effort traffic.

A congestion control mechanism for Broadband Wireless Access (BWA) networks is proposed by [76]. Based on game theoretic approach [77] which basically controls the congestion with the help of dynamic pricing and focusing on providing both the QoS and fairness aspects. The research variables or QoS metric used for this research is throughput, fairness, queuing delay and packet losses. This work can be extended regarding the real time and non real time traffic service differentiation as well as improving the congestion control mechanism. Similarly, [77] also proposed a game theoretic approach but mainly focusing on the joint solution for congestion control and power control.

Resource Management

Optimizing network resources and efficiently assigning/allocating resources is a challenging task. For the wireless systems where time-varying channels are concerned, the resource allocation can take advantage of varying channel conditions to enhance the performance of the system. [37] provides a resource allocation mechanism for real time and non real time traffic in the Orthogonal Frequency Division Multiple Access (OFDMA) networks. The QoS metrics used include: number of users, throughput, transmission rate and packet drop rate.

Alternatively, [78] provides an analytical model and analyzes the existing framework for the Wireless Personal Area Networks (WPANs). It carries extensive performance analysis of proposed solution using both network simulations and analytical results in terms of delay, blocking probability, offered load and system utilization.

Routing

Routing can also play an important role for Quality of Service specially for multimedia traffic. Not only congestion does affect the quality of Voice over IP (VoIP) calls, but protocol updates can also greatly impact real time traffic. In this context, [79] present the experimental results for checking the effect of Border Gateway Protocol (BGP) updates on the VoIP calls. The study consider following metric voice codecs, packet losses, BGP updates and average Mean Opinion Score (MOS). It states that there is a strong correlation between the intelligibility of VoIP and BGP updates. Conversely, [80] provides an algorithm for the routers to meet the delay and rate guarantees for various traffic patterns. It tries to provides better results in terms of service time, number of queued calls and end to end delay.

3.1.2 Scope of the Preliminary Study

Extensive content analysis has been conducted to get the direction and glimpse of what is going on the subject area Multimedia QoS/QoE. In this preliminary research, the articles have been selected from leading good reputed journals and conferences for meta-analysis. A total of ”210” articles have been found in such conferences and journals pertaining to Multimedia QoS/QoE from the time period of 2006–2013.

Table 2 presents the list of selected top ranked journals, conference, magazines and letters for this study. All the articles have been chosen, reviewed, critiqued, analyzed and extensive content analysis has been carried out.

The content analysis research methodology is very well defined by [41] below:

> “A method of analysis in which text (notes) is systematically examined by identifying and grouping themes and coding, classifying and developing categories”

Table 3 depicts the frequency of the selected articles from the relevant leading journals or conferences etc.

Table 2: Selected Journals/Conferences/Magazines/Letters

Name of Publisher
IEEE Transaction on Networking
IEEE Transactions on Information Theory
IEEE Communication Magazine
IEEE Journal on Selected Areas in Communications
IEEE Transaction on Multimedia
IEEE Transactions on Mobile Computing
IEEE Communication Letters
IEEE Transactions on Wireless Communications
IEEE Transaction on Communications
ACM SigComm
ACM MobiComm
IEEE InfoComm
IEEE Globecom

Various categories of research methodologies and research models have been defined by [41]. The research methodology refers to the procedure of conducting the research, whereas, the research model refers to the kind of illustration that has been presented for each articles proposed solution.

Delimitation

The major delimitation for this preliminary meta analysis is the scope of the study. Only 210 research articles have been studied and examined for meta-analysis, but the selected articles are from the top layer journals, conferences, magazine and letters. Therefore, such a collection can be considered as a strong suite for this study, and the results of this study can be strongly considered validated due to the selection of leading publications.

Table 3: Scope Of The Preliminary Study

S No	Name of Publisher	%. Of Articles
1	Globecom	23 %
2	IEEE Transactions on Multimedia	20%
3	IEEE Transactions on Wireless Communications	9%
4	IEEE Transactions on Mobile Computing	8%
5	IEEE Transactions on Networking	7%
6	IEEE Transactions on Communications	7%
7	IEEE Journal on Selected Areas in Communications	7%
8	IEEE Communication Magazine	5%
9	IEEE Infocom	4%
10	IEEE Communication Letters 6	3%
11	ACM SigComm	2%
12	ACM MobiComm	3%
13	IEEE Transactions on Information Theory	1 %

3.2 Quantitative Analysis of the proposed field

The goal of this section is to present a quantification of the importance attributed to the previously reviewed topics in the scientific literature of the past eight years. To this end, we utilized the papers (journals and conferences) previously discussed and also investigated analysis and simulation instruments employed by researchers.

Various directions are explored in the literature for multimedia quality provisioning, as listed in Table 1. Fig. 9 shows the amount of research work done in each of the subject areas defined above. The fields of Resource Management (allocation/reservation) and packet scheduling appear to be prominent. Instead, error control, mobility management, power control, routing and security have received minor attention with

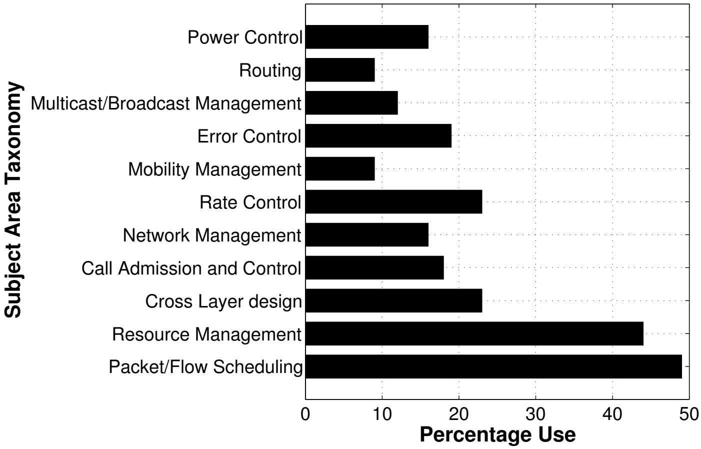


Figure 9: Usage Frequencies Of The Subject Areas Taxonomy

respect to multimedia quality provisioning.

Several research methodologies have been adopted to investigate the aforementioned subjects. According to the classification of [41], research methodologies can be subdivided into: simulation study, mathematical analysis, practical/laboratory testbed, framework description, or field experiments. These research methodologies are defined as follows

Mathematical analysis— An analytical (e.g., formulaic, econometric or optimization model) or a descriptive model is developed for the phenomenon under study.

Framework description— Research that intends to develop a framework or a conceptual model.

Simulation— Network simulation is a technique where a program models the behavior of a network either by calculating the interaction between the different network entities.

Practical/Laboratory testbed— Research in a simulated lab-

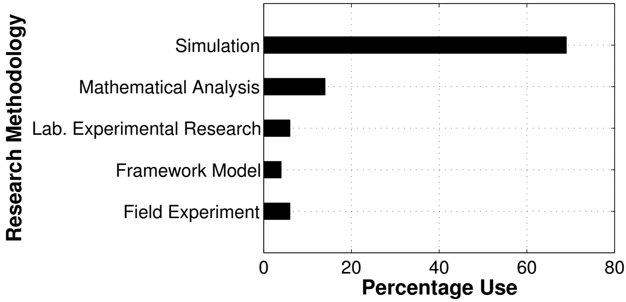


Figure 10: Research Methodologies Usage

oratory environment that manipulates and controls the various experimental variables and subjects.

Field Experiment— Research in organizational setting that manipulates and controls the various experimental variables and subjects.

The shares of each research methodology for the considered articles is reported in Fig. 10. It is highlighted how most (69%) of the research work regarding multimedia quality provisioning is based on simulation. The second-preferred research methodology is mathematical analysis, which is adopted by (13%) of the researchers. However, due to the limitation of infrastructure technologies for R&D purposes, there are very few researches that use experimentations or real-life evaluations.

Given the high percentage of studies adopting simulation as a research methodology (either as the only one, or as one of many), it may be worth investigating which kind of simulation instruments are actually used. However, collecting data about simulation tool usage from all the articles is not always easy, since unfortunately most of the researches do not provide information about the tool or simulation environment used. Although we believe that this practice should be discouraged, still we observe that about (44%) of the papers do not give to the reader enough details about the adopted simulation environment.

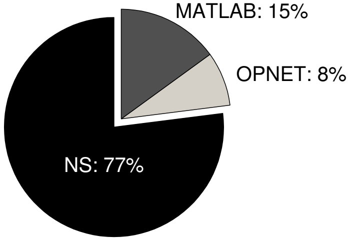


Figure 11: Simulation Environment/Tools Usage

From those who mention some details, still there is a considerable fraction of papers that report to have employed a simulator developed on-purpose by the authors. An exact evaluation about how much this allows the results to be reproducible by other researchers is probably out of the scope of the present analysis. Nevertheless, we believe that a standard simulator has the undoubtable advantage to enable comparison of the results from different contributions. In light of this reasonings, a classification of the kind of instruments used, has been reported in Fig. 11, where we sorted the papers employing

their own developed simulator according to the programming environment they used. From this figure, it is visible that the largest share of papers, among those that declare the used instruments, is represented by those employing the Network Simulator (either the NS-2 version or the newer NS-3) [81].

Instead, Fig. 12 reports the frequency of the OSI layers which are considered in the analyzed papers. The classification has been done accordingly to the OSI standard by considering physical, data link, network, transport, and application layers (presentation and session layers have been neglected, as usually done, they have 0 entries anyway). From the figure, it results that most of the investigations on multimedia quality provisioning involve the data link layer, with (46%) of the papers. Besides this layer, application and physical layers, both are the focus of (15

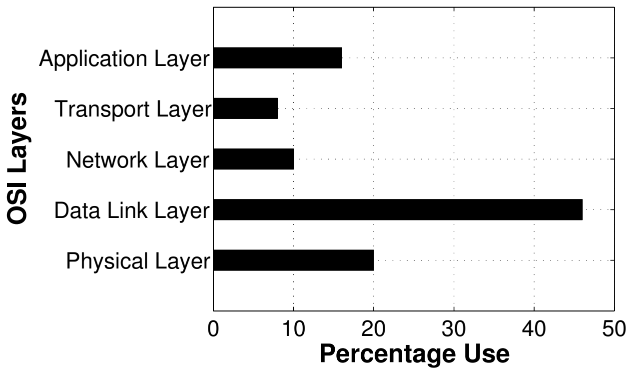


Figure 12: OSI Layer Usage

and 20%) of the considered papers, respectively.

Two points are actually worth remarking: The entire classification, and the high frequency of papers investigating layer 2 may be due to the common presence of cross-layer investigations, which therefore involve more than one OSI layer. Therefore, from these numbers one may infer that cross-layer analysis involving physical layers and medium access control (seen as a part of the data link layer) is fairly common. However, it is also surprising that little work deals with intermediate layers, i.e., network and especially transport layers (only 8% of the papers).

Another interesting classification can be done for what concerns the underlying technology. Due to the proliferation of standards and communication technologies, the specific technological supports, on which multimedia communications are considered, are very variegate and heterogeneous. For the sake of classification, we consider here a subdivision mostly driven by medium access control criteria, as this has been shown in Fig. 12 to be the most explored layer. Therefore we distinguish among the papers considering Internet-like architectures, also including Local and Wide Area Networks based on the same medium access. For what concerns wireless networks, we distinguish between coordinated access, which either refer to cellular networks, or to other deterministic multiplexing (e.g., OFDMA based), and contention-based access, which mostly involve WLANs, e.g., based on collision avoidance paradigms. It is worth noticing that many papers consider the integration among different technologies; in the case of heterogeneous wireless access the most challenging aspects still resides in the lack of guarantee (or the possibility of providing only soft QoS) in the collision-based techniques. Therefore, heterogeneous networks can be considered as a subset of contention-based access. Finally, we consider sensor network as a separate item here, which may be marginal to multimedia communications, although their importance is expected to rise in the close future.

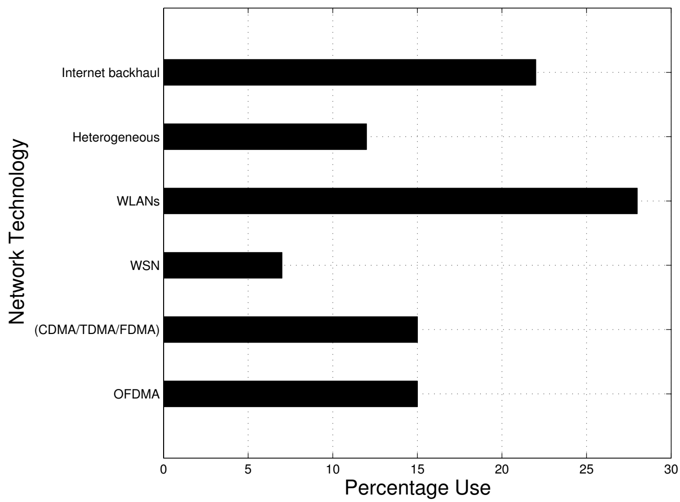


Figure 13: Underlying Network Technology Usage

The results reported in Fig. 13 show that about one third of the research has been performed over contention-based technologies, i.e., (29%) of the cases. However, a large share of papers, (22%), focus on Internet-like technologies; observing that (12%) of the papers consider heterogeneous networks, we may remark that an even larger share than that of Internet-like techniques explores the issue of quality provisioning for WLAN or similar techniques, possibly in combination with others. As expected, the relevance of wireless sensor networks is still limited, only (7%) of the papers. Yet, we argue that this value is already significant, given the technological impairments of Wireless Sensor Networks (WSNs) compared to other scenarios, and it may be expected that this fraction will become even larger, as the papers exploring multimedia over sensor networks are all relatively recent. Finally, coordinated access is investigated in (15%) of the papers; it is worth mentioning that, although OFDMA-based access techniques represent a significative example of such technologies, only (2%) of the papers specifically explore *Long Term Evolution* (LTE) networks. Of course, we expect this value to increase dramatically in the next years together with further deployment of LTE networks.

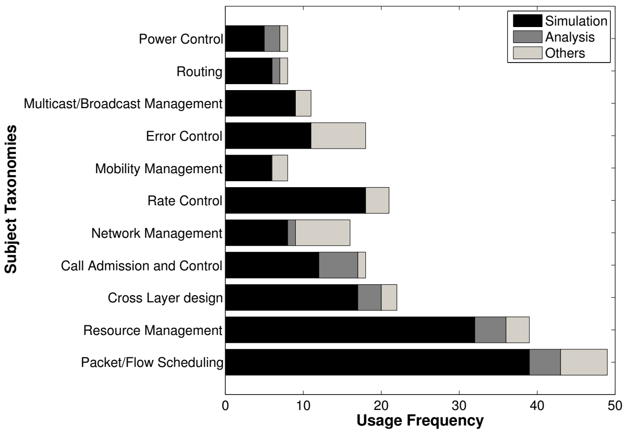


Figure 14: Research Methodologies vs. Subject Taxonomies

Fig. 14 reports the methodology adopted to derive the results for each of the taxonomy subjects. Although the simulative approaches are dominant, it is possible to observe a relevant presence of mathematical models in the areas of scheduling, call admission control and cross-layer design; these are mostly optimization frameworks, which are solved to derive theoretical upper bounds (often impractical, as they are computationally expensive), and are in a majority of the cases further confirmed by means of simulation. Scheduling and network management (and to some extent, also routing and error control, although to a lesser extent given their lower general frequency) sometimes adopts practical testbeds and on-field evaluations as well. It is worth remarking that other subjects, such as multicast/broadcast and rate control, in spite of their critical role with respect to multimedia application, are almost entirely investigated by means of simulation.

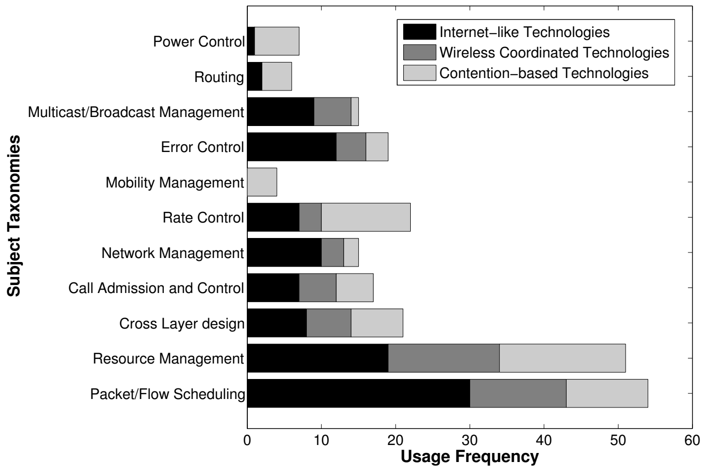


Figure 15: Network Technology vs. Subject Taxonomies

Fig. 15 discusses the subject areas versus the kinds of network technology they relate to. For the sake of compactness in the figure, we grouped the network technologies in three macro-groups, namely Internet-like wireline architectures, coordinated wireless access (involving deterministic multiplexing, cellular networks, and so on, i.e., anywhere a single centralized authoritative unit is present to perform resource allocation), and on the other hand contention-based wireless access (including WLANs, sensor networks, heterogeneous scenarios, where access collisions may happen). Scheduling is an important subject for all the three classes, but wireline scenarios are also mostly investigated for routing, network management, and multicast; conversely, coordinated networks are the most important scenario, although not the only one, for resource management strategies, which is meaningful thinking of the case of cellular networks. Contention-based wireless access are considered especially for cross-layer design, call admission control, rate control and mobility management.

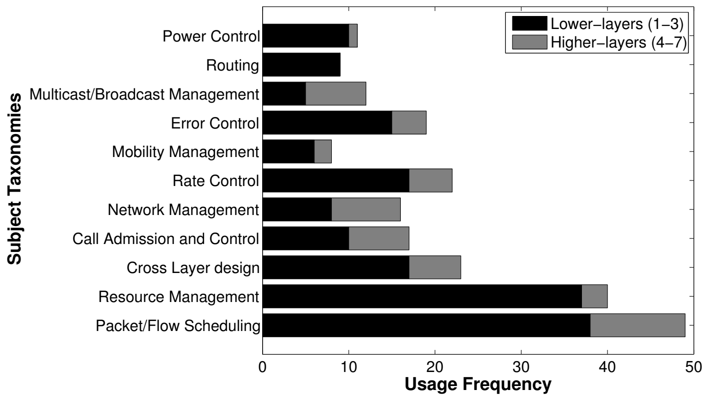


Figure 16: OSI layer vs. Subject Taxonomies

The different research areas are also investigated in Fig. 16 for what concerns the OSI layers they involve. Again for compactness reasons, we just limited the split to lower (1-3) and higher (4-7) layers. Note that, according to Fig. 12, intermediate layers are investigated less frequently. Not surprisingly, topics like power control and routing concern for the most the lower layers, while network management and call admission control also see a relevant share of application-focused studies. For multicast/broadcast, the application layer is even predominant. For other topics such as error control and scheduling we see an interesting split, i.e., about (70%) and (30%) of the papers consider lower and higher layer solutions, respectively. In all likelihood, the point here is that the general concept of "scheduling" or "error control" is applied in an extremely different manner between the data link layer and the transport/application layers. In reality, in spite of a similar terminology, the proposed solutions may be extremely different due to layer-specific approaches.

Finally, one point that is not so encouraging about the current state of the art research is that cross-layer design almost entirely focuses on lower layers. Although it is not directly visible from the figure, the reason is mainly that "cross-layer" almost always mean channel-state dependent allocation, i.e., to keep into account physical layer conditions when performing radio resource management. Surely, this is an important issue which is correctly investigated by many researchers, even though it should not necessarily be specific of multimedia quality provisioning, and has been discussed about since long before the explosion of video services. Nevertheless, we believe that also the cross-layer interactions with higher layers, i.e., keeping into account the unique characteristics of video traffic, as well as transport layer aspects of delay versus rate requirements or the impact of video caching should be worth of more attention. In this sense, the research community should, in our opinion, encourage a larger effort towards this direction.

Chapter 4

Multimedia Scheduling and Resource Management

In future 4G communication networks, it is expected that most of the bandwidth will be used to serve multimedia applications. Thus, there is a need for scheduling mechanisms, which can manage various types of multimedia communication (interactive, conversation, video streaming, etc.) and provide adequate Quality of Experience according to the application needs. The most promising candidate for 4G systems, i.e., Long Term Evolution, also integrates Femtocells as a cost-effective solution for pervasive communication. In such a scenario, the implementation of effective scheduling mechanisms becomes even more important. Finally, before implementing scheduling policies on real devices, validation through simulation studies is often employed. For all these reasons, we investigated the implementation of an adaptive scheduling mechanism for an LTE scenario with Femtocells within the well known NS-3 simulator.

On the other hand, one of the reference applications to support real-time traffic is the Real-time Transport Protocol (RTP), which can be used to transmit multimedia contents on real-time basis. At the same time, Real-time Transmission Control Protocol (RTCP) is used for receiving feedback and getting information about the network. This work proposes and evaluates a traffic management implementation in such an RTP/RTCP environment for congestion control. Deficit Round Robin queue discipline is used as the traffic management strategy instead of Random Early Detection and DropTail queue disciplines. A simulation campaign was performed to analyze the effects of implemented traffic strategies in RTP/RTCP environment and compare it with previous solutions. The obtained results highlight a significant difference in terms of jitter delay and packet losses and improvement for the bandwidth utilization for real-time flows. Thus, we are able to provide quantitative evidence of the importance of the queue discipline to efficiently manage multimedia content.

4.1 Design of a Unified Multimedia-Aware Framework for Resource Allocation in LTE Femtocells

4G systems are often identified with the Long Term Evolution (LTE) of 3G cellular systems, such as the Universal Mobile Telecommunications System (UMTS). LTE offers multi-carrier approach for multiple access. For the downlink, it uses Orthogonal Frequency-Division Multiple Access (OFDMA) and Single-Carrier Frequency-Division Multiple Access (SC-FDMA) in the uplink direction. LTE incorporates the evolution of radio access with the help of Evolved-UTRAN (E-UTRAN), therefore, LTE in conjunction with System Architecture Evolution, which includes Evolved Packet Core, is known as Evolved Packet System (EPS).

EPS uses the concept of bearer, which is basically the flow for traffic with defined Quality of Service (QoS) requirements for the User Equipment (UE). According to 3GPP standards, the QoS parameters for LTE systems are: QoS Class Identifier (QCI), Allocation/Retention Priority (ARP), Guaranteed Bit Rate (GBR), Maximum Bit Rate (MBR) and Available Maximum Bit Rate (AMBR). The QCI involves the following parameters for each bearer type: Resource type, Priority, Packet Error Loss Rate and Packet Delay Budget. Scheduling of radio bearers is done in eNodeB

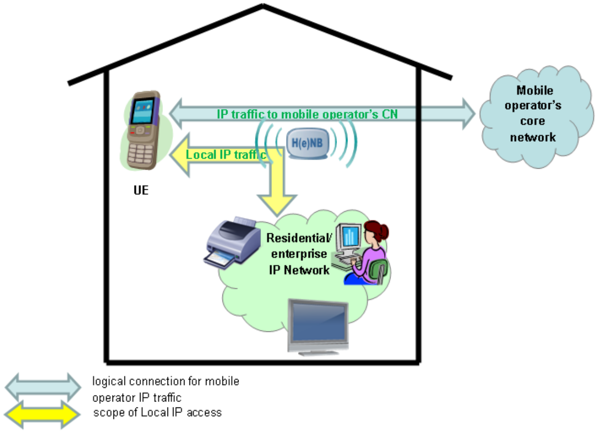


Figure 17: Generic Femto Network Architecture

of E-UTRAN, so that resources can be allocated according to their QoS requirements and availability in eNodeB. The eNodeB is the intermediate point between all the users and core network; therefore, radio resource management takes place at the eNodeB.

It is expected that future communication networks will have more and more real time contents. As surveyed by Cisco in 2011 [3], this will amount to 90% of traffic by 2016. In such a context, it will become more common to translate from QoS to Quality of Experience (QoE), i.e., also involving subjective perception of delivered content. At the same time, increased demand of traffic and coverage is leading to low-cost solutions for network deployment. One promising solution in this sense is represented by femtocells. A femtocell employs coordinated access technology, which is low powered with reduced cost, and can be used to enhance the cell capacity/coverage as well as provide self-organization features. The generic femtocell scenario is illustrated in Fig. 17, [82].

The equipment that is in the customer-premises and provides connectivity with a 3GPP UE over E-UTRAN wireless air interface to the mobile operator network using a broadband IP backhaul is known as Home eNode B (HeNB) in [83]. Another interesting point to exploit LTE

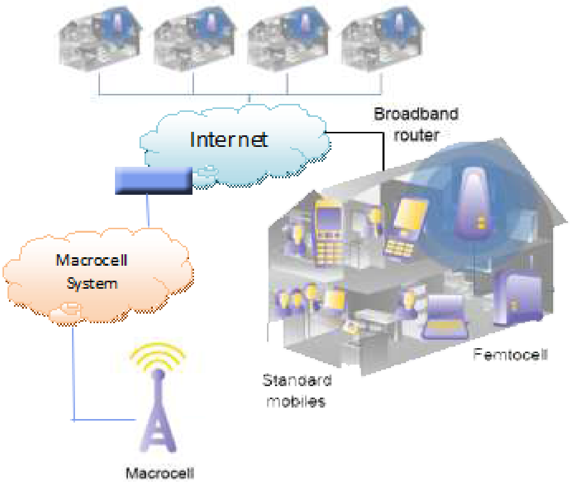


Figure 18: Femtocells Traffic Flow

femtocells can be traffic offloading, because local contents will not pass through core network or the eNB. These can be scheduled and routed exclusively via the HeNB, as in Fig. 18.

Referring to our report [84], where we analyzed latest research trends for Multimedia QoE/QoS provisioning, by considering collected articles from top leading journals, conferences, magazines, and letters in the last 5 years, we found that Radio Resource Management (RRM) aimed at multimedia is still relatively unexplored in LTE networks. Even though researchers investigated Multimedia QoS for other OFDMA-based technologies, for example IEEE 802.16, very little attention has been paid to multimedia service provisioning in LTE networks, especially including femtocells. This is the major motivation behind this research.

RRM is an important research issue in high speed communication networks which are facing an increasing demand for high quality service, in particular for multimedia applications. Optimizing network resource allocation and efficiently assigning resources is a challenging task. For wireless systems, where communication channels are time-varying, the resource allocation can take advantage of variable conditions to enhance the system performance. An important issue for future communication systems is quality provisioning relatively to multimedia applications, which often have demanding requirements. Particularly, video packets have stringent deadlines and are prone to packet losses [85]. Moreover, the RRM needs also to address the cases where a mixture of multimedia and non-real time traffic is present.

Among the existing solutions where real time and non real time traffic are managed over an OFDMA network, we mention [37] that provides a resource allocation mechanism involving the following QoS metrics: number of supported users, throughput, transmission rate and packet drop rate. Also [78] considers different kinds of traffic, but the technology of reference is a Wireless Personal Area Network (WPAN). This paper presents an extensive performance evaluation, by means of both network simulation and theoretical analysis, to get results in terms of delay, blocking probability, offered load and system utilization.

In particular, this work focuses on the Downlink Scheduling in HeNB. The goal here is to provide better quality to the end users which are running GBR applications, while at the same time maintaining an acceptable quality for non-GBR users, in the following generically referred to as NGBR. To this end, we implemented a simple priority algorithm which will be referred to as Adaptive Fair Delay Prioritized Scheduling (AFDPS) and is described in detail in the following sections. We have implemented AFDPS within the Network Simulator 3 (NS-3) [81], and verified the applicability of such a solution to provide adequate QoE level to both GBR and NGBR users.

4.1.1 Proposed Adaptive Scheduling Mechanism

The main objective of the present paper is to provide a unified framework for downlink scheduling and radio resource management for LTE network. The proposed solution is developed for the base station of an LTE femtocell known as HeNB. The purpose of this algorithm is to provide efficiency to GBR contents while at the same time preserving fair-

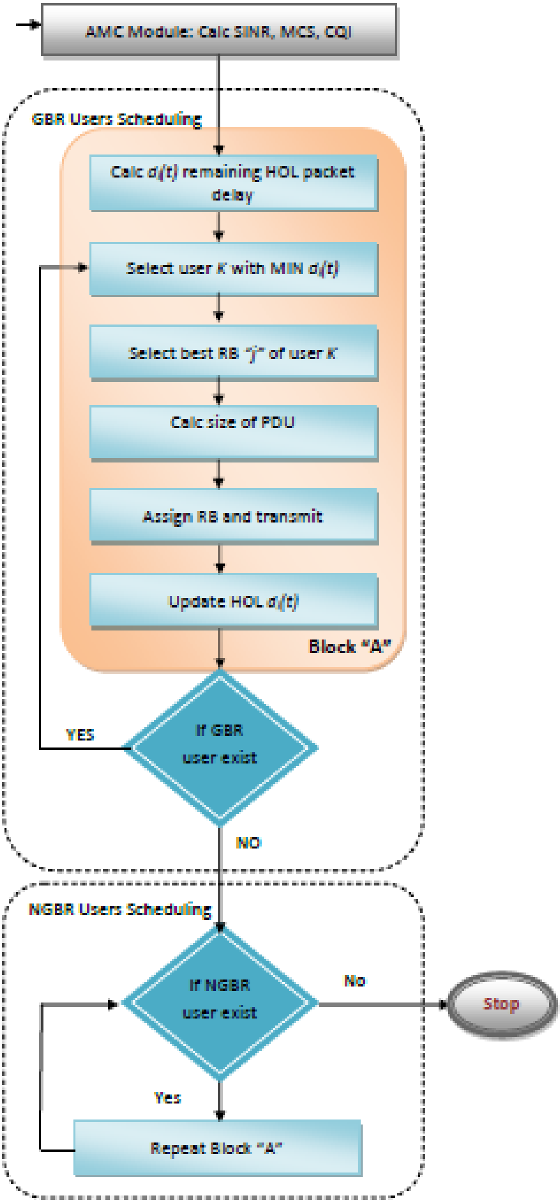


Figure 19: Algorithm Workflow

ness for NGBR contents as well.

The proposed model is a priority based scheduling mechanism, where GBR traffic users are prioritized over NGBR ones. For both classes of users, a channel-aware scheduling rationale is applied. The flow diagram of the proposed model is shown in Fig. 19. This algorithm is basically adaptive and opportunistic, in a sense that it assigns the Resource Blocks (RBs) to the users based on their Channel Quality Indicator (CQI) value, which changes over time. Thus, the overall resource block assignment changes accordingly. In the downlink, the SINR is calculated for each RB assigned to data transmissions which is the ratio of the power of the intended signal from the considered eNB by the sum of the noise power plus all the transmissions on the same RB coming from other eNBs (the interference signals).

Thus, this algorithm acts at the MAC layer by taking advantage of information from the physical layer (PHY) and utilize it at the Radio Link Control (RLC) layer, particularly within the scheduler. This enhances the efficiency of resource management process to be dynamic and opportunistic.

First of all, the RLC layer calculates the Head Of Line (HOL) delay for the bearers, and then it passes this value to the MAC scheduler. The scheduler calculates the remaining HOL delay according to the current time and then it re-arranges all the bearers in ascending order according to their remaining HOL delay values. This sorting is performed so that urgent bearers, which have minimum remaining HOL delay, get RBs assigned earlier and choose the initial best ones. In this way, the GBR bearers get better QoS. The RBs allocation is done in Proportional Fair manner [86]. Before assigning RBs to the selected user, the size of the Packet Data Unit (PDU) is computed based upon already collected CQI and computed MCS value for those users. Therefore, the size of the PDU might be different from that of the upper layer (SDU) , because it depends on the channel condition for that user. In this sense, this algorithm is opportunistic/adaptive in the packet size as well. Therefore, the computations regarding the queue size, SDU arrival time, HOL delay are crucial and should be done carefully. Once the RBs are assigned to selected users in the scheduler, the RLC module is triggered to transmit the specified amount of data, i.e., the computed PDU size, from each bearer. After the PDU is transmitted, the HOL delay is updated, and the resulting value is sent from the RLC to the scheduler at each scheduling instance, i.e., every millisecond.

Similarly, after GBR bearers, the scheduler assigns resources to NGBR bearers, analogously to what done for GBR contents. In this way, it also provides efficiency for NGBR bearers, but comparatively GBR bearers resource allocation will get better QoE, as it takes place before NGBR resource allocation. This prioritization actually depends on the operator policy concerning the allocation of resources between GBR and NGBR bearers at each scheduling instance. Other implementation choices are possible according to policy of the network operator.

4.1.2 Implementation

The entire framework has been implemented and evaluated within NS-3 [81], in particular using the LTE module developed by the LENA project [87] which supports the LTE MAC Scheduler API defined by the Femto Forum [82]. The ns-3 simulator is open source and can model different kinds of communication networks. It offers the advantage of being modular and describes the protocol stack in a comprehensive manner. In particular, the extension proposed in [87] models LTE networks in a detailed manner, especially for data link and physical layers. Our goal is to extend this representation so as to include cross-layer interactions with upper layers, especially the transport layer, where multimedia traffic may require special handling.

In particular, the structure of ns3 for LTE includes four modules which can be implemented with interchangeable solutions. The first one, called Radio Resource Control (RRC), acts a container of bearers with specified QoS requirements. The second one, modeling the RLC layer, associates the bearers with the physical devices. Two further modules, denoted as MAC and PHY, model the medium access and the physical channel, respectively. In particular, for the LTE implementation, the PHY layer provides a CQI measurement to all the associated UEs, which enables channel aware scheduling. Our solution deals with the MAC module and was developed with reference to the MAC scheduler interface defined in the Femto Forum API [88] for what concerns the interactions with the other entities.

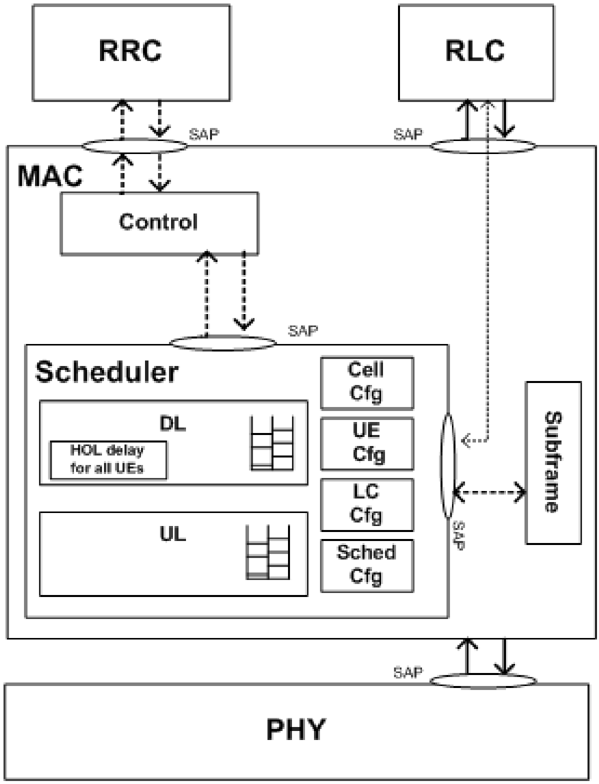


Figure 20: MAC Scheduler Interfaces

Fig. 20 shows the flow of data and control between the MAC scheduler and PHY, RLC, and RRC. The primitives are also defined in the API, which are passed through various Service Access Points (SAPs). At every Transmit Time Interval (TTI) the subframe within the MAC triggers the scheduler. The TTI is set according to 3GPP standards as 1 millisecond.

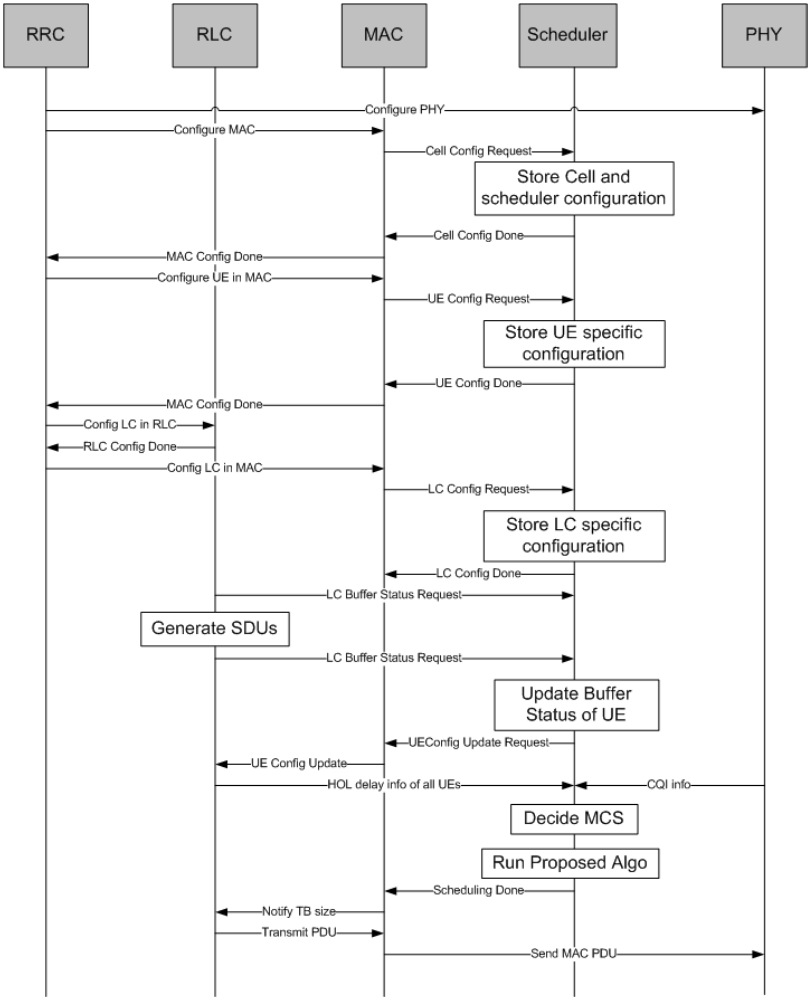


Figure 21: Sequence Diagram of System

The sequence diagram of the initial setup starting from RRC until PHY is represented in Fig. 21. In particular, the original routines that our implementation develops are pictured in the "Scheduler" block interacting with the MAC entity of the ns3 simulator. Importantly, the developed software is entirely modular, and fully compatible with any design choice made in the other blocks, the only cross-layer requirements being the availability of a CQI value at the PHY module and the presence of a SDU-packetizer in the RLC module that enables the queue length updates in the scheduler, as visible from Fig. 21 under the "RLC" block.

We detailed the implementation of a multimedia-aware LTE scheduler for femtocell scenarios. The scheduler manages two priority classes, for GBR and NGBR traffic. This was successfully implemented in the well-known NS-3 simulator. The resulting framework is entirely modular and therefore transparent to any specific choice in the simulation modules, as well as in the network entities it interacts with. Moreover, it can be adapted to different design needs for what concerns the management of multimedia flows.

4.2 Evaluation of Deficit Round Robin Queue Discipline for Real-time Traffic Management in an RTP/RTCP Environment

The work investigates transport layer solutions for real-time delivery, especially focusing on the choice of the queue discipline for multimedia flows. In this sense, it is important not only to achieve high efficiency of the queueing policy, but also to be able to correctly manage different kinds of traffic. In fact, multimedia traffic comprises several applications with different characteristics in terms of required QoS. Moreover, it is expected that multimedia traffic will coexist with other best effort data traffic in the same network operations.

Technological solutions to achieve real-time delivery over the internet include in particular the Realtime Transmission Protocol (RTP) and the Real-Time Control Protocol (RTCP) [89]. In [90], RTP/RTCP environment was introduced and implemented within the well-known ns-2 simulator [81].

In this paper, we proceed along the lines of [90], that is, the main functionality of RTP is modeled as involving the identification of payload, the generation of RTP packets, and finally the introduction of RTP packets time stamps and sequence numbers. RTCP is used for inquiring the network status and getting feedback. The major advantage of RTCP is that it does not interact with RTP, but it can be used as a network management entity. Thus, RTP and RTCP can operate jointly as direct and feedback loop.

The data exchanged by the nodes through this mechanism enter a buffer queue at each intermediate receiver. One basic cause of delay in the transmission of multimedia traffic is actually the queueing delay at these buffers. Thus, when multimedia or real-time traffic is concerned, it is important to select the correct type of queueing policy in order to provide the users with the required QoS. To this end, different choices are possible. Previous existing work utilizes very simple queue disciplines, such as a basic DropTail policy [91].

When real time flows are considered, it is extremely important to monitor the traffic and performing dynamic resource allocation, as argued by [92]. In that paper, the authors investigate an RTP/RTCP environment and use the queueing delay parameter to tune the congestion control mechanism. However, they do not focus on a specific queue discipline; rather their approach tries to determine whether to increase the packet train by observing the current queueing delay. We take instead a similar approach but we focus on the queue discipline choice. Moreover, we take also the delay jitter into account. In fact, as described by [93], high values of the jitter are caused by the network congestion and/or inadequate queue disciplines. Therefore, selecting the proper queue discipline can play a tremendous role in the efficiency of real-time traffic over congested network.

In this work, we propose to use, within the RTP/RTCP framework, a Deficit Round Robin (DRR) strategy. This is justified by several theoretical benefits, which we aim at validating in practice. To assess the validity of this approach, we implemented this queue policy within the ns-2 simulator and we quantify its performance in a test topology by means of a simulative campaign, evaluating several metrics of interest. In this way, we are able to verify that the proposed solution is properly able to fulfill

QoS requirements of multimedia traffic.

4.2.1 The RTP/RTCP Environment and the Proposed Solution

RTP was developed by the Audio-Video Transport Working Group. It uses regimented packet format for multimedia contents that is, audio and video [89]. It was designed for multicast applications. RTP provides the following services: Identification of payload, time stamping, sequence numbering, and delivery notification. It provides end-to-end network transport functions, but is not responsible for guaranteed QoS. Therefore, RTP and RTCP are most frequently used in a joint manner, because RTP is used to transport multimedia data and RTCP is employed for monitoring the network QoS [94].

For multimedia sources, adaptive transmission rate algorithm is introduced in [95] using a TCP friendly rationale. The algorithm considers the maximum transmission rate, minimum transmission rate and the granularity. In the adaptive algorithm, the sender changes its transmission rate according to the adaptive algorithm schema. The authors of [95] have simulated TCP-friendly and constrained TCP-friendly flow control and their results proved that the constrained TCP-friendly version reaches a higher degree of fairness than the plain TCP-friendly one. This combination of RTP for real-time flows and RTCP to monitor the QoS of the network was also used in [96], where new feedback control mechanism for video transmissions are presented. By means of simulation, this contribution compares the packet losses of UDP flow against the UDP_RTP flow. Results proved that UDP_RTP can improve the video transmission.

RTCP was also used in [97] to get the network information and tune the system accordingly for real-time traffic. But the emphasis of this paper is on multimedia traffic management in ad hoc networks. Authors are also using RTCP for getting end-to-end feedback information about the packet loss performance as well as delay jitter. RTP/RTCP protocol suite is well known for getting feedback from the receiver and it is also utilized by [98] to exemplify the real-time traffic flow for unicast and multicast environment. In the present analysis, we are also focusing on multicast scenario, particularly for real-time traffic. The authors of [98] present a vast survey of multimedia synchronization and the main technique is exemplified by the existing RTP/RTCP suite. Congestion Control is a challenging issue particularly for real-time traffic, this argument is also supported by [99], where however it is proposed to use Random Early Detection (RED) queueing discipline for congestion control or for dropping the packet at the time of congestion. However, this may violate the need for real-time traffic to receive fair bandwidth allocation and especially the requirement for lossless delivery of information.

Therefore, we also take a RTP/RTCP environment as the starting point of our evaluation. However, as will be argued in the next sections, we stress the importance of an efficient queueing discipline at the buffers. For this reason, in the following we review and discuss this point. Further, we comparatively evaluate different choices in this respect.

The simplest queue discipline, called DropTail, follows a very basic policy, i.e., it treats all the packets equally in a single queue, and each packet is served in the same order as received; that is, this is simply a First-In First-Out (FIFO) approach which adopts the dropping rule that all the packets exceeding the buffer capability are discarded when the buffer is full. This approach involves very low computational complexity and easily predictable behavior [91]. However, it also causes significant drawbacks, i.e., increased delay, jitter and packet losses for real-time applications. Further, entire bursts of packets might be discarded and a significant queueing delay may increase the overall network congestion.

The buffer management scheme used in previous RTP/RTCP framework, such as [90], is simply to employ Droptail at the sender and receivers ends, whereas RED queue discipline is used at the router ends. The RED queueing policy uses priority levels to drop packets; however, since the RED queue is applied at the link between the two routers, therefore it treats all the flows identically. Actually, RTP flows should be given some priority over non real-time flows. Therefore, considering the jitter and packet losses as QoS metrics, jitter was introduced due to the employment of Droptail and packet loss was increased due to the packet dropping by the RED queue. Alternatively, traffic management strategy can be applied to this scenario for real-time flows, so that real-time can be given better QoS over non real-time traffic. This will result in enhancing the QoS in terms of increased share of bandwidth and decreased jitter and packet losses for real-time flows. In the proposed traffic management strategy, Deficit Round Robin (DRR) is used as a queue discipline for the source and destinations of real-time flows as well as for the link between the routers.

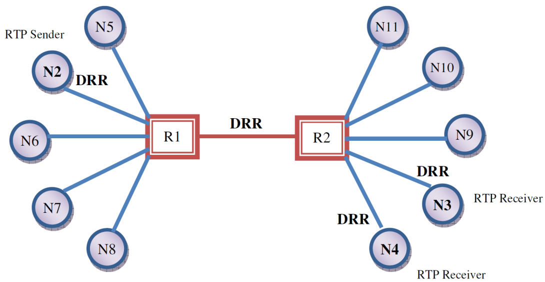


Figure 22: Evaluation Network Topology

The network topology is described in Fig. 22. In the above figure, N2 is the node 2, which the sender of RTP flows, R1 and R2 are the edge routers and N3 and N4 are the receiver nodes. Instead of simple Drop-Tail strategy, we have implemented a DRR queue discipline; in this manner, the jitter delay and the packet loss rate are significantly decreased, whereas the bandwidth utilization for real-time flows is improved. Results have been proven through simulation, described in the following section.

4.2.2 Performance Evaluation

We utilized the NS-2 environment [81] to perform a simulation campaign. Focusing on a topology as previously described, we implemented the DRR policy for its use at the queueing buffers. The RTP sender and receiver use the DRR queue, similarly, DRR is also used instead of RED queueing system, but for all the other nodes Droptail is used. The queue length is set to 50 packets for each of the queue, whereas for the DRR queue between the routers has the capacity of 100 packets. Further, RTP-RTCP multicast environment is arranged in such a way that node 2, node 3 and node 4 join the multicast group, where node 2 is the transmitter of RTP traffic whereas node-3, node-4 are the RTP traffic receiver.

We simulated 100 seconds of transmissions and we evaluated the resulting system performance in terms of three different metrics: bandwidth allocation, jitter, and smooth loss. We compare our results with those reported in [90], where DropTail and RED traffic management strategy was adopted; thus, the curves referring to this approach are labeled "without traffic management." We will show instead that in our proposed solution, buffer management plays very important role in terms of bandwidth allocation and other traffic management aspects; for this reason, we refer to our solution as "with traffic management" in the graphs. The following subsections detail the analysis for each of the investigated metrics.

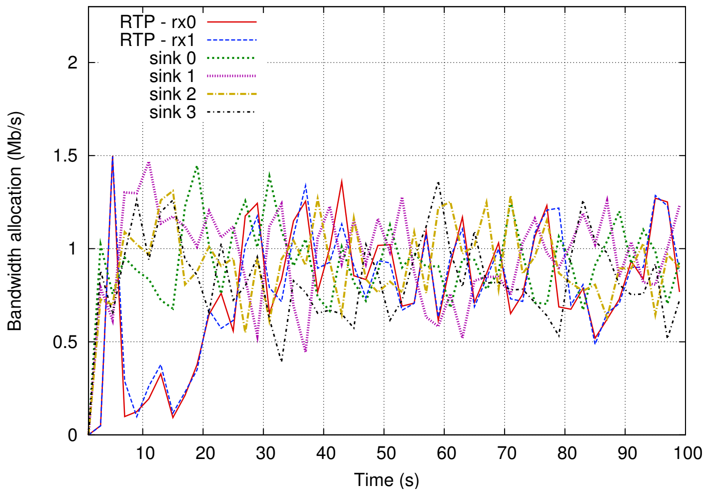


Figure 23: Bandwidth Allocated To The Flows Without Traffic Management

Bandwidth Allocation — Figs. 23 and 24 compare the bandwidth allocation of the flows, for the two cases without (previous solution) and with traffic management (proposed solution), respectively. In Fig. 23, it is observed that all the flows get almost equal (and not very high) amount of bandwidth. No preference is given to real-time flows. Finally, the bandwidth allocations of all flows fluctuate considerably, and in a very variable manner from flow to flow.

Conversely, when we apply our proposed traffic management strategy, there is a great difference of bandwidth consumption between the real-time flows and the others. Fig. 24 shows that the two RTP flows get a higher amount of bandwidth than what reported in Fig. 23, and also they receive a better allocation than the other flows, which is a sign that real-time traffic is correctly provided with better QoS than best effort traffic. It is also worth observing that the oscillations are significantly reduced. Still, they are not avoided; observe, for example, that the bandwidth assigned to real-time flows drops in many points, due to congestion. Yet, the plots of Fig. 24 highlight a more regular behavior, that is,

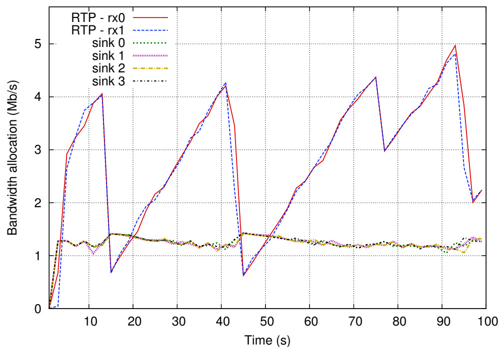


Figure 24: Bandwidth Consumptions Of The Flows With Traffic Management

similar kinds of flow enjoy similar QoS at the same time.

Figure 25: Jitter Analysis

Jitter — Thanks to its better traffic management capability, our proposed strategy is also able to effectively decrease the jitter values as compared to the previous solution with Droptail and RED. Fig. 25 shows a comparison of the jitter values for multimedia nodes between the solution of [90] and new proposed solution with traffic management for RTP/RTCP environments. We distinguish jitter values for our proposed strategy “with traffic management” and the implementation of [90] which is referred to as “without traffic management.”

The plot emphasizes that jitter is significantly decreased when our traffic management technique is applied on RTP/RTCP environment. It is also important to notice that the jitter values decrease and become smoother as time goes by, but still the curve of the proposed queue discipline stays considerably below the one without traffic management features.

Smooth Loss — Real-time traffic is very sensitive and requires more priority as compared to non real-time flows. Thus, by applying some traffic management strategies, real-time traffic can be given precedence over non real-time. Thus, when real-time traffic is prioritized, packet losses are likely to be decreased for real-time flows. This is investigated in Fig. 26, which represents the smooth losses [100] comparison between the previous RTP/RTCP solutions with our solution containing the traffic management component. The smoothing factor, i.e., the weight coefficient of the loss rate value evaluated in the previous time instant, is set to $\alpha = 0.9$. Again we report two plots, “with” and “without” our traffic management approach.

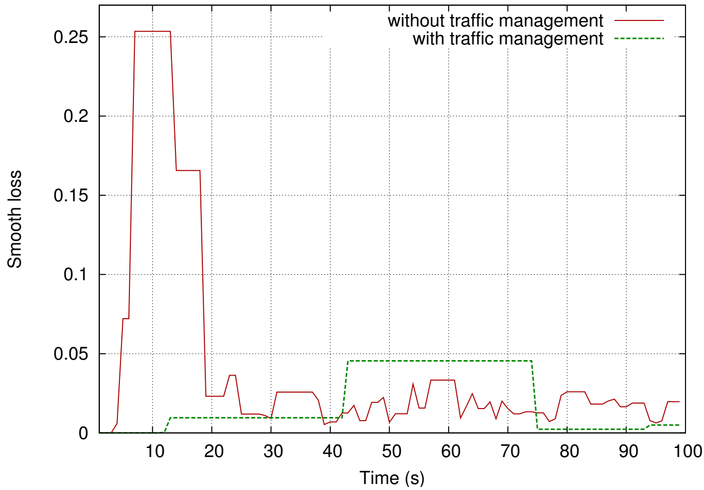


Figure 26: Smooth Loss Analysis

The considerably better behavior in terms of packet losses is expected, depending on the application in use, to reflect in an enhanced QoS for real-time flows.

We considered an RTP/RTCP environment where RTP is used to transmit multimedia and realtime data and RTCP is employed in conjunction with RTP to monitor the network statistics and maintain the overall end-to-end network QoS in a feedback-based manner. In related research work, RED queue discipline, or even simpler DropTail, are used to manage multimedia traffic. However, the adoption of different queue disciplines can improve the transmission of RTP traffic. For instance, in the considered scenario, the introduction of Deficit Round Robin significantly improves the performance in terms of bandwidth allocation, jitter, and packet losses for real-time flows.

The advantages of DRR have been proven by means of simulation results comparing existing solutions with a proposed approach based on DRR in terms of all these performance metrics. The correct choice and setup of queueing policy at the network nodes has been proven to be key for meeting QoS constraint of real-time multimedia traffic.

Possible extensions of the present work include the analysis of similar traffic management strategies in different topologies and/or with different combinations of existing queue disciplines. Moreover, it is also possible to envision the application of similar approaches to RTP/RTCP environments realized over wireless networks.

Chapter 5

Wireless Video Optimization Framework

To investigate the performance of the PHY/Application cross-layer optimization we consider an analytical model, where the channel is modeled by means of a Markov chain, whose states represent different channel qualities.

Depending on the channel quality, an efficient decision has to be made regarding packets' resource allocation, so as to provide the user with an adequate quality, which turns into maximizing the number of delivered packets and satisfying the relational requirements between layers (e.g., no need for EL packets if their corresponding BL is not decoded or received).

This problem is addressed first in a unicast scenario, then extended to a multicast scenario, where the resource allocation becomes a challenging trade-off among the different needs of the users.

In this chapter, we define an analytical model for network management policies, whose objective is to maximize the end-user quality for scalable video contents. We also define the users' taxonomy in terms of user satisfaction levels during transmission. It is also investigated that how user feedback impact the perceived video quality and how can it bring improvements for quality.

On the other hand, we also address following questions: Is PHY/ Application cross-layer optimization really required for video transmission over next generation wireless networks? Or would a sequential allocation where optimization is independently performed at the PHY and Application layers work similarly? How do cross-layer and non-cross-layer optimization perform also compared to the theoretical best allocation that one could apply, if the channel states and the user quality requirements were all known in advance? Is there a way to adapt to the channel variability? How do the unicast scenario extend to the multicast case? Given that a compromise allocation must be found between the needs of all the users in the multicast group, it may be that cross-layer optimization is insufficient.

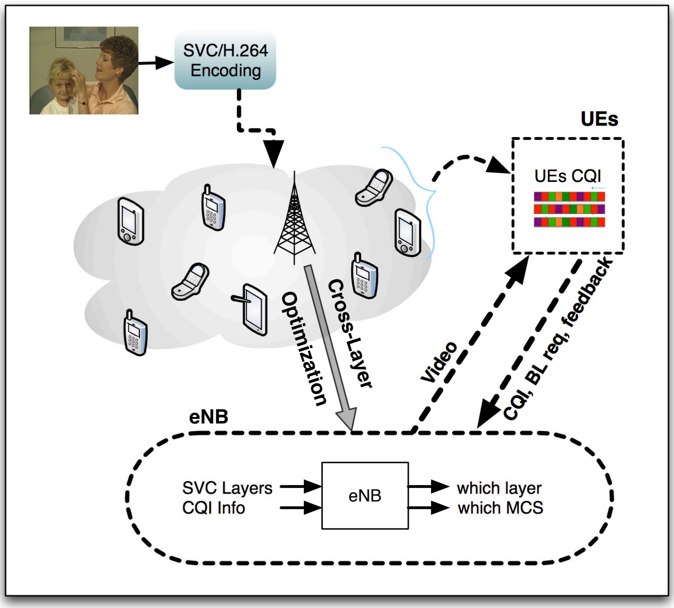


Figure 27: Generic View Of Wireless Video Transmission

5.1 Proposed Framework — WiViOpt

The proposed solution is named as **Wi**reless **Vi**deo **Opt**imality (WiViOpt), which tries to explot the cross layer features of OSI protocol stack, while optimizing the requirements of video users to provide the improved quality in wireless cellular network.

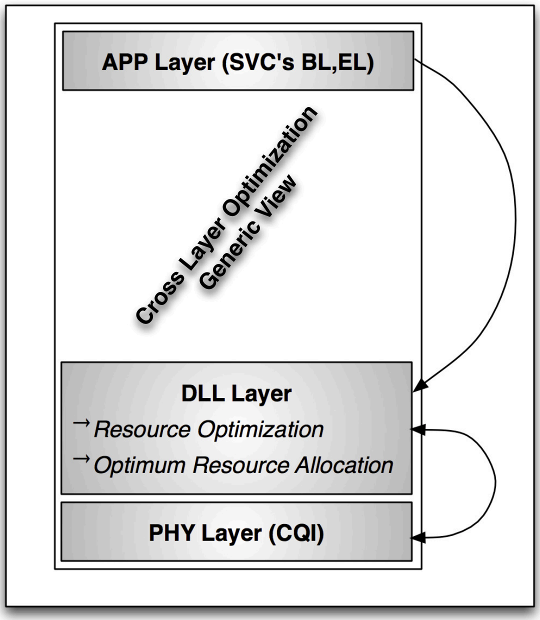


Figure 28: Cross Layer Approach

Fig. 27 represent the general setup for video optimization. It illustrates the flow of information from source to destination. In this case, there is video source which is first endcoded into SVC/H.264 by the encoder. This encoded bitstream is transmitted to the base station (eNodeB/eNB). The base station received the user channel status through Channel Quality Indicator (CQI), depending upon which the eNB checks for Modulation and Coding Scheme (MCS) on which the resources for transmission are to be allocated to the user.

Before and during the transmission, the users send feedback about the number of transmit opportunities (TXOPs) they require to correctly decode the video, because of which the eNB updates its policies and assigns optimum resources, as per aforementioned steps. In the Chapter. 6, we investigate both scenarios about having a feedack not, and compare the results.

Our motivation is to exploit the cross-layer design features to provide enhanced video quality and tune the network according to application requirements and channel conditions (through PHY layer) of the users, and then assigning optimum resources at the data link layer of OSI stack. As we study the cross-layer design, therefore, general view of our approached cross-layer design is depicted in Fig. 28.

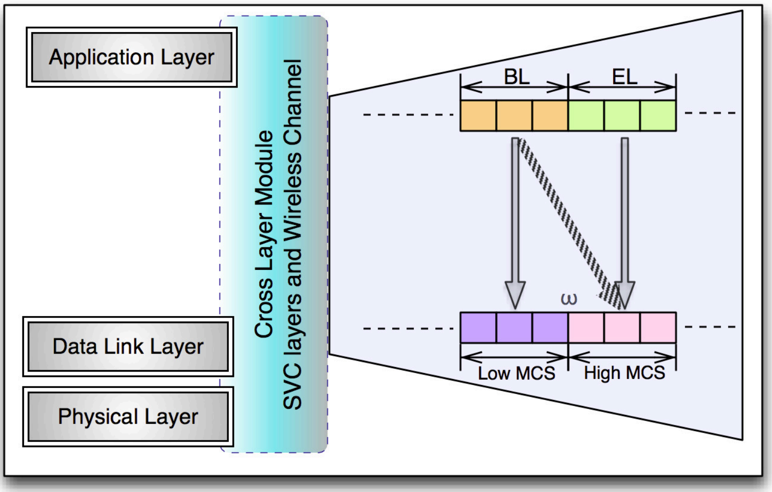


Figure 29: PHY/Application Cross-layer Allocation Concept

Each user has its own requirements for BL and EL packets. Fig. 29 illustrates the basic cross layer design for SVC exploiting adaptive chanel rates. The BL packets can be transmitted with any MCS available, but

EL packets can be transmitted with an MCS above a certain threshold ω (defined in Sec. 5.2.4).

5.2 Analytical Model

Taking advantage of adaptive channel rates and scalable video layers for quality enhancement can significantly improve the perceived quality experienced by the users. For this reason, we try to exploit both features to provide enhanced QoE for scalable video while taking into consideration the varying channel conditions of individual users. For simplicity, we just assume two layers, i.e., one enhancement layer beyond the base layer. The extension to multiple enhancement layers would be conceptually straightforward.

The relationship of BL and EL can be better understood with the help of Fig. 30, which shows how BL and EL packets are related. This is a simplified version of what reported in [101], since only one BL and one EL are considered, also only one level cross-dependance between layers. This scheme can be promptly extended to multiple layers and more sophisticated temporal dependencies. The decoding of each EL frame relies on the decoding of the corresponding BL frame. Thus, assuming for simplicity that one frame corresponds to one video packet, when a BL packet is lost, the corresponding EL is useless at the receiver, and also the decoding of the entire video flow will suffer, as some frames will be skipped. Instead, upon losing an EL packet, the receiver will still be able to decode the video flow. Hence the quality decrease would be relatively minor; in the worst case, the basic video quality of playing only the BL is still guaranteed.

Thereafter, indices "1" and "2" refer to the base and enhancement layers, respectively. The notations used throughout this work are presented in Table 4. Each user has its own requirements for BL and EL packets, and the BL requirement of each user is a uniformly distributed random variable between 40% and 70% of the total number of time slots available.

This section mathematically models the choice of the optimal number of transmit opportunities (TXOPs) required for the users to decode

Table 4: Notations and Assumptions

Paramter	Description
T	Total number of k time slots
N	Number of users $j \in N$
$\mathcal{L}, \mathcal{Z}, \mathcal{S}$	Set of SVC layers, MCS and Channel States, respectively
ζ, ℓ	Generic elements of $\mathcal{Z}$ and $\mathcal{L}$
t_1, t_2	Number of time slots for a base and enhancement layer of video, respectively; note that $t_1 + t_2 = T$
p_j, P_j^d	Packet loss and decoding probability of user j, respectively
θ_j	Threshold for base layer packets that each user j must receive to correctly decode the stream
$\hat{g}_j$	Expected goodput of user j
$\hat{n}_1, \hat{n}_2$	Requested no. of BL and EL packets, respectively
n_1, n_2	Received no. of BL and EL packets, respectively
$g^k_{\ell,\zeta}, d^k_{\ell,\zeta}, q^k_{\ell,\zeta}$	Goodput, packet delivery delay and quality with MCS ζ and SVC layer ℓ at time k, respectively
γ	Weight of the linear combination of goodput and delay
G_j, D_j, Q_j	Goodput, packet delivery delay and quality of the user j, respectively
χ_{avg}, χ_{max}	Number of BL packets computed as multicast value using Average and Maximum strategy, respectively
α, β	The weight for BL and EL quality assessment, respectively
λ	Fractions of transmissions reserved for BL
ω	Threshold of channel quality, below which EL cannot be transmitted reliably

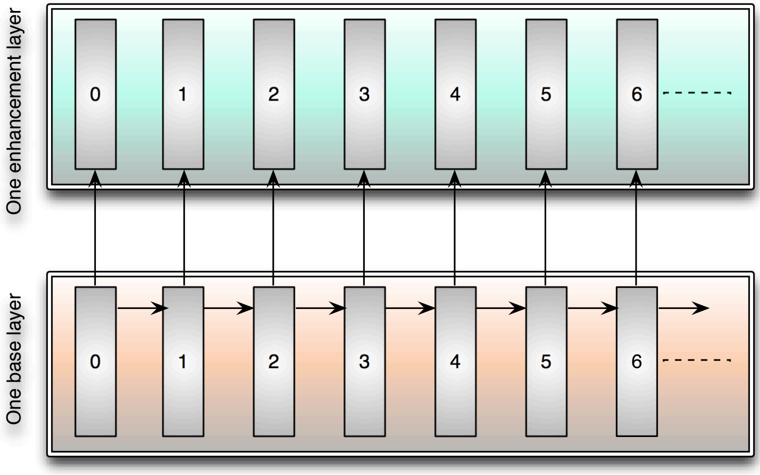


Figure 30: Temporal And Quality Dependencies Between BL and EL.

the base layer and maximize their perceived quality in unicast and multicast environment, and it also introduces a taxonomy of user categories according to their satisfaction level. We consider a fixed number of TXOPs, since this supposedly model the real time characterization of video (after T TXOPs, we start with new video contents periodically). Finally, the framework also describes the network policies evaluated in subsequent section.

5.2.1 Taxonomy of Users

To categorize the user satisfaction about the perceived video quality, we define the following user taxonomy:

Satisfied — Within the current batch of video content transmission, a user j is said to be *satisfied*, if it has received enough base layer packets to decode the video within available number of transmit opportunities, i.e., $n_1(j) > \theta_j$.

Happy — A user is said to be *happy*, if it expects that it will eventually receive the required number of base layer packets to decode the video within the available transmit opportunities. Whenever this expectation is confirmed, this user will become satisfied. Conversely, if the expectation is contradicted, the user will become unhappy or hopeless.

UnHappy — A user is said to be *unhappy*, if it will likely not receive the required number of base layer packets to decode the video before the end of the available transmit opportunities. Similarly, confirmation of this expectation leads the user to become *Hopeless* or the expectation can change to *Happy* or *Satisfied*.

Hopeless — User j is said to be *Hopeless* if it is sure that it will never be satisfied within the available number of transmit opportunities. That is, it will not receive the required number of base layer video packets to decode correctly within the available transmit opportunities, i.e., $n_1(j)(T) < \theta_j$.

5.2.2 Mathematical Model Scenarios

We define the analytical model for optimizing the number of transmit opportunities required to correctly decode the BL for each user, the mathematical model considers two scenarios. In the former, we consider a single packet transmission per time slot and a single user in the system. This scenario is useful to understand the latter, where multiple packets are transmitted depending on the channel quality, and also multiple users are considered in unicast as well as multicast environment.

In both scenarios, we consider the PHY/Application Cross-Layer Optimization of SVC video over heterogeneous channel conditions with the help of adaptive modulation and coding schemes. We also study the impact of having a feedback from the user, optimizing the quality of video and analyzing satisfaction level of the users.

Unicast Scenario — Initially, we focus on a single user j trying to decode the video stream (or, alternatively, we can consider several users identical to each other). We assume that different packets are decoded with independent identically distributed (i.i.d) probabilities for the sake of simplicity. Therefore, the probability to correctly receive a packet is $1 - p_j$. The decoding probability of the flow for user j corresponds to the probability of decoding θ_j packets out of t_1 TXOPs, i.e.

$$P_j^d = \begin{cases} 0 & \text{if } (1-p_j) \cdot t_1 < \theta_j \\ F & \text{if } (1-p_j) \cdot t_1 \geq \theta_j \end{cases} \tag{5.1}$$

where F is defined as

$$F = \sum_{k=0}^{t_1-\theta_j} \binom{t_1}{k} (1-p_j)^{t_1-k} (p_j)^k \tag{5.2}$$

We formalize the expected goodput of user j as:

$$\hat{q}_j = \begin{cases} 0 & \text{if } \textit{non-decoding} \\ \alpha \cdot \hat{n}_1(j)(1-p_j) + \beta \cdot \hat{n}_2(j)(1-p_j) & \text{if } \textit{decoding} \end{cases} \tag{5.3}$$

where α and β are properly chosen weights that regulate the relative importance of base layer packets versus enhancement layer packets in terms of supplied quality, therefore, $\beta > \alpha$ to give more weight to EL. Thus, the average perceived goodput is

$$\begin{aligned} \mathbb{E}[\hat{q}_j] &= \Big(\alpha t_1(1-p_j) + \beta t_2(1-p_j)\Big) \\ &\times \sum_{k=0}^{t_1-\theta_j} \binom{t_1}{k} (1-p_j)^{t_1-k} (p_j)^k u[t_1-k] \end{aligned} \tag{5.4}$$

where $u[x]$ is a unit step, i.e., it is one if x is non-negative, zero otherwise.

It is straightforward to identify the optimum value of t_1 that maximizes the expected quality as per (5.4) from an a priori standpoint. Therefore, we define $Opt_{t_1}(j)$ as the value of t_1 between 0 and T that provides maximum expected quality for user j. Even if we have multiple users, but all have identical error probability p_j, it is immediate to derive the best policy for the operator, i.e., the optimal split of T into t_1 and t_2 so that the quality according to (5.3) is maximized. Thus, formally

$$Opt_{t_1}(j) = \arg\max \mathbb{E}[q_j] \tag{5.5}$$

subject to

$$\begin{aligned}(1-p_j)\cdot t_1 &\geq \theta_j\\ T &= t_1+t_2\end{aligned}$$

We can write the former constraint as $t_1 \longrightarrow p_j \cdot t_1 + \theta_j = g$, and later one as $t_1 + t_2 - T = h$ and name it as h. Further, if the goal function to be maximized is denoted as f, we have the following function and constraints:

$$\begin{aligned}f(t_1,t_2,p_j) &= \Big(\alpha t_1(1-p_j)+\beta t_2(1-p_j)\Big) \\ &\times \sum_{k=0}^{t_1-\theta_j}\binom{t_1}{k}(1-p_j)^{t_1-k}(p_j)^k u[t_1-k] \qquad (5.6)\\ g(t_1,p_j,T) &= \theta_j + p_j\cdot t_1 \qquad (5.7)\\ h(t_1,p_j,T) &= t_1+t_2-T \qquad (5.8)\end{aligned}$$

The above formulated problem can be solved by utilizing the Lagrange Multiplier optimization method to find out the maxima [102], so that we can find the additional number of TXOPs required to correctly decode the video and end user quality is maximized. According to Lagrange Multiplier, to solve this optimization problem, we need to take partial derivative of $f(t_1,t_2,p_j)$, $g(t_1,t_2,p_j)$ and $h(t_1,t_2,p_j)$ with respect to t_1, t_2 and p_j.

Rewriting f to simplify the expression

$$f(t_1,t_2,p_j) = \Big(\alpha t_1(1-p_j)+\beta t_2(1-p_j)\Big)\cdot\mathcal{D}(p_j,t_1) \qquad (5.9)$$

To solve this formulated problem, we need to take partial derivative of $f(t_1,t_2,p_j)$, $g(t_1,t_2,p_j)$ and $h(t_1,t_2,p_j)$ with respect to t_1, t_2 and p_j. Therefore, first of all, taking partial derivative of f (5.9) with respect to t_1, t_2 and p_j.

Once, we have found partial derivatives of f, g and h, now we need to put Lagrange multipliers, the Ψ with the partial derivatives of the g and

μ with the partial derivatives of the h. Then we can put the respective partial derivatives of f, g and h equivalent, as follows:

$$\begin{aligned}
\frac{\partial}{\partial t_1, t_2, p_j} f &= \frac{\partial}{\partial t_1, t_2, p_j} g \cdot \Psi + \frac{\partial}{\partial t_1, t_2, p_j} h \cdot \mu \\
\frac{\partial}{\partial t_1} f(t_1, t_2, p_j) &\longleftrightarrow \frac{\partial}{\partial t_1} g(t_1, t_2, p_j) \cdot \Psi + \frac{\partial}{\partial t_1} h(t_1, t_2, p_j) \cdot \mu \\
\frac{\partial}{\partial t_2} f(t_1, t_2, p_j) &\longleftrightarrow \frac{\partial}{\partial t_2} g(t_1, t_2, p_j) \cdot \Psi + \frac{\partial}{\partial t_2} h(t_1, t_2, p_j) \cdot \mu \\
\frac{\partial}{\partial p_j} f(t_1, t_2, p_j) &\longleftrightarrow \frac{\partial}{\partial p_j} g(t_1, t_2, p_j) \cdot \Psi + \frac{\partial}{\partial p_j} h(t_1, t_2, p_j) \cdot \mu
\end{aligned}$$

$$\alpha(1 - p_j) = p_j \Psi + \mu \tag{5.10}$$

$$\beta(1 - p_j) = 0 + \mu \tag{5.11}$$

$$-(\alpha t_1 + \beta t_2) = t_1 \Psi + 0 \tag{5.12}$$

Using (5.10), (5.11) and (5.12), we can solve and get the values of t_1 which will provide the optimum value on which expected quality is maximized. Therefore, from (5.11) $\mu = \beta(1 - p_j)$, put the value of μ in (5.10)

$$\begin{aligned}
\alpha(1 - p_j) &= p_j \Psi + \beta(1 - p_j) \\
(\alpha - \beta)(1 - p_j) &= p_j \Psi \\
\Psi &= \frac{(\alpha - \beta)(1 - p_j)}{p_j}
\end{aligned}$$

Putting this value of Ψ in (5.12), we get

$$
\begin{aligned}
-(\alpha t_1 + \beta t_2) &= t_1\Big(\frac{(\alpha-\beta)(1-p_j)}{p_j}\Big) \\
-\alpha t_1 p_j - \beta t_2 p_j &= \alpha t_1 - \beta t_1 - \alpha t_1 p_j + \beta t_1 p_j \\
\beta t_1 - \alpha t_1 &= \beta t_1 p_j + \beta t_2 p_j \\
t_1(\beta - \alpha) &= \beta p_j (t_1 + t_2) \\
t_1 &= \frac{\beta p_j (t_1 + t_2)}{(\beta - \alpha)} \\
t_1 &= \frac{\beta p_j T}{(\beta - \alpha)}
\end{aligned}
$$

Now the optimum number of transmit opportunities would equal to θ_j plus the additional number of transmit opportunities as in (5.13) required to successfully decode the video. Therefore, lets label it as:

$$
\eta = \frac{\beta p_j T}{(\beta - \alpha)} \tag{5.13}
$$

Now the optimum t_1 is:

$$
Opt_{t_1} = \theta_j + \eta \tag{5.14}
$$

We further expand the analysis to a scenario in which each user can receive from 0 to n packets at each transmit opportunity, according to their loss probability.

We assume the number of received packets is Bernoulli distributed, related to the individual loss probability. Based on loss probability p_j, n probabilities are calculated, $\{\rho_0(j), \rho_1(j), \rho_2(j) \,...\, \rho_n(j)\}$. Therefore, at each TXOP, depending on its loss probability, user j expects to receive 0 to ν_j packets, where

$$
\nu_j = \sum_{i=0}^{n} i\rho_i(j)
$$

The decoding probability P_j^d becomes

$$P_j^d = \begin{cases} 0 & \text{if } \nu_j \; t_1 < \theta_j \\ q_j & \text{if } \nu_j \; t_1 \geq \theta_j \end{cases} \tag{5.15}$$

where q_j is the expected quality for user j, defined as

$$\begin{aligned} \mathbb{E}[q_j] &= \alpha\nu_j t_1 + \beta\nu_j t_2 \\ &= \alpha\nu_j t_1 + \beta\nu_j (T - t_1) \end{aligned} \tag{5.16}$$

Similar to (5.5), we can find out the optimum value of t_1 satisfying, $\nu_j t_1 > \theta_j$ and $t_1 < T$, which provides the maximum expected quality given P_j^d for user j:

$$Opt_{t_1}(j) \quad = \quad \arg\max \mathbb{E}[q_j] \tag{5.17}$$

In unicast, the base station allocates resources according to the individual needs, therefore, χ_j, for each user j, is the minimum threshold, so that after which various policies (XLO/NXLO, described in sec. 5.2.4) play role, which can be defined as

$$\chi_j = Opt_{t_1}(j) \times \lambda \tag{5.18}$$

where $\lambda \in [0, 1]$ is a parameter to compute the threshold χ_j of the base layer, that is, the amount of base layer, which must be received by the receiver. For example, $\lambda = 0.5$ means we restrict the system to transmit 50% of the BL packets, after that the system has the choice to transmit BL or EL packets depending on the user's need and channel quality.

Note that in general each user will have a different optimal value of t_1, that is each user j has its own Opt_{t_1}, therefore the operator needs to find a suitable tradeoff among the heterogeneous needs of all the served users for multicast session.

Multicast Scenario — Once each user (UE) has requested its required number of transmit opportunities to the central base station (eNB), and the individual optimal t_1 of each user is found, then based on various strategies, the base station decides optimum number of transmit opportunities to transmit BL and EL for all the users in multicast group. In this work, we take into consideration following two strategies. Note that in the multicast transmission, the base station has to consider the individual channel condition of all the users, and consider aggregate channel condition for the transmission of the same video contents to all the users.

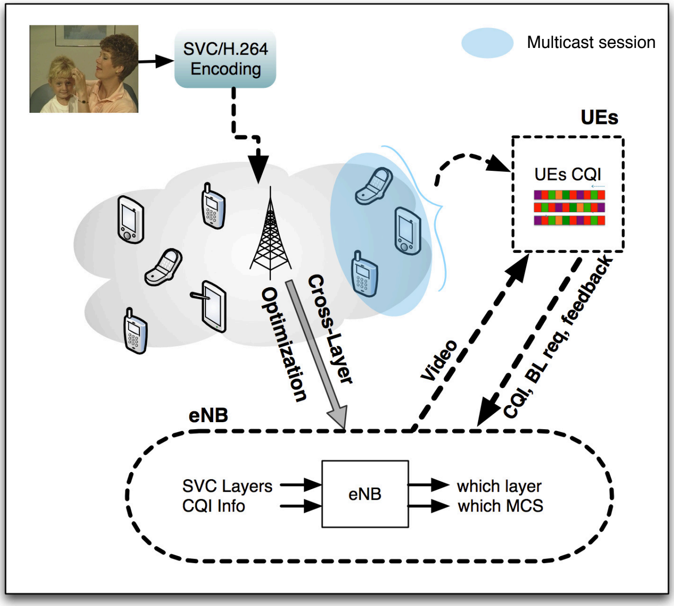


Figure 31: Multicast Scenario

Average Strategy — In this case, the base station computes the average $Avg\_Opt_{t_1}$ of all values $Opt_{t_1}(j)$, and then transmits χ_{avg} packets from the base layer, the rest are enhancement layer packets. Therefore, for all the users, based on (5.14), the optimum amount of timeslots for BL transmission value will be computed as follows:

$$Avg\_Opt_{t_1} = \frac{1}{N}\sum_{j=1}^{N} Opt_{t_1}(j) \qquad (5.19)$$

Some users will be served with fewer base layer packets than their requested $\hat{n}_1$, and some of them can even be unable to decode the stream. This strategy focuses on the quality improvements. Similar to unicast case, the minimum threshold (regarding the number of BL received packets) to decode video in multicast group based on χ_{avg}, can be defined as

$$\chi_{avg} = Avg\_Opt_{t_1} \times \lambda \tag{5.20}$$

Maximum Strategy — The users declare their preferred $\hat{n}_1$ and the central base station takes a conservative approach, selecting the largest $\hat{n}_1$ as the maximum of the declared values, to satisfy all the users. With this strategy, all the users may reasonably expect to be satisfied in the end, thus it tries to focus on fairness for all users. Therefore, at the beginning of the transmission they are all *happy* as defined in 5.2.1. However, this strategy may decrease the QoE because more base layer and fewer enhanced layer packets will be received by all the users.

$$Max\_Opt_{t_1} = \arg\max_{j \in N} \Big(Opt_{t_1}(j) \Big) \tag{5.21}$$

The minimum threshold (regarding the number of BL received packets) to decode video in multicast group based on χ_{avg}, can be defined as

$$\chi_{max} = Max\_Opt_{t_1} \times \lambda \qquad \exists \;\; n_1(j) \geq \chi_j \quad \forall j \in N \tag{5.22}$$

5.2.3 Resource Allocation Model

The wireless channel is modeled as a Markov chain, with the set of transition probabilities. We consider a set of states, $\mathcal{S} = \{\mathcal{S}_1, \mathcal{S}_2, \mathcal{S}_3, \cdots, \mathcal{S}_s\}$, where each state is associated to a channel quality level in increasing order. Denoting the transition from state i to state j as $p_{i \to j}$, the transition probabilities can be collected into a matrix

$$\mathbf{P} = \begin{bmatrix} p_{1\to1} & p_{1\to2} & p_{1\to3} & \cdots & p_{1\to(s-1)} & p_{1\to s} \\ p_{2\to1} & p_{2\to2} & p_{2\to3} & \cdots & p_{2\to(s-1)} & p_{2\to s} \\ p_{3\to1} & p_{3\to2} & p_{3\to3} & \cdots & p_{3\to(s-1)} & p_{3\to s} \\ \vdots & \vdots & \vdots & \vdots\vdots\vdots & \vdots & \vdots \\ p_{s\to1} & p_{s\to2} & p_{s\to3} & \cdots & p_{s\to(s-1)} & p_{s\to s} \end{bmatrix}.$$

Each possible state in $\mathcal{S}$ is associated to a level of robustness and quality of the channel. Depending upon the quality of channel, the transmission rate can be adapted according to the Adaptive Modulation and Coding schemes available, that is, using different MCS for different states. Therefore, we have a set of MCSs, $\mathcal{Z} = \{\zeta_1, \zeta_2, \zeta_3, \cdots, \zeta_z\}$. Similarly, as mentioned before, taking the advantage of scalable video, which consists of one base layer and multiple enhancement layers, therefore, we have a set of SVC layers, $\mathcal{L} = \{\ell_1, \ell_2, \ell_3, \cdots, \ell_l\}$. We associate each time slot to an index k.

The goodput of a user can now be defined as:

$$G_j|\mathcal{S}_{\ell\zeta}^k = g_\ell^k(j) \tag{5.23}$$

Given the channel state $\mathcal{S}_{\ell\zeta}^k$, at time k, the user is able to receive packets from layer ℓ with channel rate ζ. The number of base layer packets, n_1, to be decoded correctly in order to play the video is under the condition that $n_1 \geq \theta_j$. In the subsequent sections, the meaning of θ is modified according to the scenario. Thus, we can define the overall system goodput as follows:

$$Q = \frac{1}{N} \sum_{j=1}^{N} \sum_{k=1}^{T} G_j|\mathcal{S}_{\ell\zeta}^k \tag{5.24}$$

The delivery delay definition is:

$$D_j|\mathcal{S}_{\ell\zeta}^k = d_{\ell\zeta}^k(j) \tag{5.25}$$

Similarly, the overall system delivery delay can be defined as:

$$D = \frac{1}{N} \sum_{j=1}^{N} \sum_{k=1}^{T} D_j|\mathcal{S}_{\ell\zeta}^k \tag{5.26}$$

We can define a performance metric for user j, at time slot k, which takes into account the impact on the perceived quality of goodput and delivery delay, such as:

$$Q_j|\mathcal{S}_{\ell\zeta}^k = \gamma \cdot g_{\ell\zeta}^k(j) - (1-\gamma) \cdot [d_{\ell\zeta}^k(j)]^{-1} \tag{5.27}$$

where $\gamma \in$ [0,1] is a tunable parameter. Thus, the overall system quality is defined as:

$$Q = \frac{1}{N} \sum_{j=1}^{N} \sum_{k=1}^{T} Q_j | \mathcal{S}_{\ell\zeta}^{k} \tag{5.28}$$

5.2.4 Allocation Policies with Algorithms

We design three allocation policies: (i) a theoretical upper bound on the performance as using a genie-like channel knowledge and offline optimization, denoted as *Offline*; (ii) a sequential selection with optimizations performed separately at PHY and application layer, denoted as *NXLO*; (iii) a joint cross-layer optimization of both PHY and application layers, denoted as *XLO*.

```
Input: (𝓛, 𝒵)
Output: Optimum assignment of BL/EL
for each user j ∈ N do
    for each time slot k ∈ T do
        if ζ_j|S^k_{ℓζ}(j) ≥ ω and n_2(j) < t_2(j) then
            transmit EL with ζ_j;
            n_2(j) ← n_2(j) + 1 ;
        else
            transmit BL with ζ_j;
            n_1(j) ← n_1(j) + 1 ;
        end
    end
    for each time slot k ∈ T do
        if ζ_j|S^k_{ℓζ}(j) < ω and n_1(j) < χ_j then
            transmit BL with ζ_j;
            n_1(j) ← n_1(j) + 1 ;
        end
    end
end
```

Algorithm 1: Offline Policy

```
Input: (ℒ, 𝒵)
Output: Optimum assignment of BL/EL
for each user j ∈ N do
    for each time slot k ∈ T do
        if n_1(j) ≥ θ_j then
            BL_index(j) ← 0;
        end
        if BL_index(j) == 0 then
            if n_2(j) < n_1(j) then
                if ζ_j|S^k_{ℓζ}(j) ≥ ω and n_2(j) < t_2(j) then
                    transmit EL with higher ζ_j ;
                    n_2(j) ← n_2(j) + 1 ;
                end
            else
                if ζ_j|S^k_{ℓζ}(j) < ω and n_1(j) < χ_j then
                    transmit BL with lower ζ_j;
                    n_1(j) ← n_1(j) + 1 ;
                end
            end
        else
            if ζ_j|S^k_{ℓζ}(j) < ω and n_1(j) < χ_j then
                transmit BL with lower ζ_j;
                n_1(j) ← n_1(j) + 1 ;
            end
        end
    end
end
```

Algorithm 2: NXLO Policy

Offline Policy

Under this policy, the complete evolution of the channel is known a priori and is used as upper bound for comparison. This policy takes into consideration the slots with best available channel conditions amongst $\mathcal{Z}$ above a certain channel quality threshold, ω, for EL and remaining slots for BL, upon the condition that the required BL packets have been transmitted.

If the remaining slots are below a certain threshold ω, then only BL packets can be transmitted with either high or low MCS, while EL cannot, as shown in Algo. 1. Furthermore, if the BL is completed and the only available slots are below ω, then the EL packets are dropped and no more transmissions are performed.

```
Input: (𝓛, 𝒵)
Output: Optimum assignment of BL/EL
for each user j ∈ N do
    for each time slot k ∈ TT do
        if n_1(j) ≥ θ_j then
            BL_index(j) ← 0 ;
        end
        if BL_index(j) == 0 then
            if ζ_j|S^k_{ℓζ}(j) ≥ ω and n_2(j) < t_2(j) then
                transmit EL with higher ζ_j ;
                n_2(j) ← n_2(j) + 1 ;
            else
                transmit BL with higher ζ_j;
                n_1(j) ← n_1(j) + 1 ;
            end
            if ζ_j|S^k_{ℓζ}(j) < ω and n_1(j) < χ_j then
                transmit BL with lower ζ_j;
                n_1(j) ← n_1(j) + 1 ;
            end
        else
            if ζ_j|S^k_{ℓζ}(j) ≥ ω and n_1(j) < χ_j then
                transmit BL with higher ζ_j;
                n_1(j) ← n_1(j) + 1 ;
            else
                transmit BL with lower ζ_j;
                n_1(j) ← n_1(j) + 1 ;
            end
        end
    end
end
```

Algorithm 3: XLO Policy

NXLO

In the non-cross-layer policy the base station first picks the SVC layer packet to be sent, that is, BL or EL packet based on χ_j (for unicast) and χ_{avg} (for multicast). Then it checks the channel status of the user in the current time slot. Before reaching the χ_j or χ_{avg}, the system is forced to transmit the BL packets, so that the video can be correctly decoded at the receiver end with the minimum number of BL packets. The procedure is illustrated in Algo. 2.

XLO

In this policy the channel status is checked first, then the base station jointly selects the video packet (BL/EL) to be sent based on χ_j or χ_{avg}, for unicast or multicast scenarios, respectively. Similarly, before reaching the χ_j or χ_{avg}, the system is forced to transmit the BL packets until the minimum sufficient number of BL packets is collected to correctly play the video, as described in Algo. 3.

Chapter 6

Numerical Results

In this chapter, we first describe the generic scenarios for unicast and multicast video transmissions, and then we extend these scenarios to investigate the presence of feedback and its impact on perceived video quality as well as user satisfaction levels. Additionally, we also describe the resource allocation scenario for unicast and multicast video transmission. After defining all the scenarios, we discuss the simulation results for all the defined scenarios in terms of perceived quality for various strategies, as defined in Section 5.2. We further discuss the results for Cross-Layer optimization versus non-Cross-Layer optimization techniques and the impact of a unicast and multicast transmission for scalable video goodput, packet delivery delay and quality.

The performance of the aforementioned policies is assessed by means of simulation in Matlab and C++. H.264/SVC is transmitted from the central base station (hereafter denoted by eNB) to all the end users (hereafter denoted by UE(s)).

The uniform random distribution of all UEs is illustrated in Fig. 32, it shows that UEs have good and bad channel conditions, therefore, some UEs can exploit higher modulation and coding schemes, whereas, other UEs might have lower MCS due to bad channel conditions.

Each UE receives an H.264/SVC video within T time slots. Based on varying channel conditions over time and on the adopted scheduling

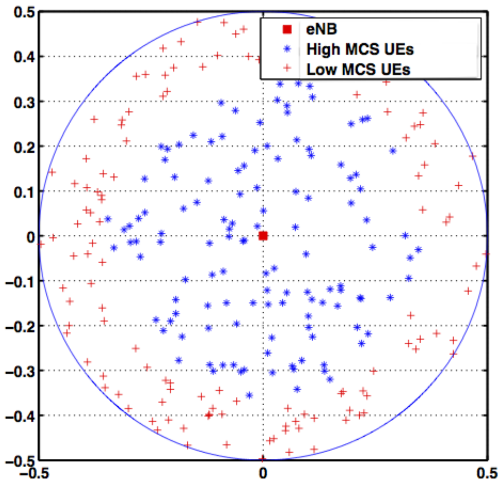


Figure 32: Differentiating UEs Based On MCS — A Generic-Distribution View

policy, each UE will get a certain amount of BL and EL packets. Multiple UEs receive unicast video streams in parallel. This is coordinated by the eNB when the same video is broadcast to all UEs in multicast scenario. Table 5 presents the simulation parameters used (if not specified otherwise)

6.1 Scenarios Description

In general we investigate all the policies and strategies in two main scenarios, that is, unicast and multicast video transmission from eNB to all the UEs. These two scenarios are further divided into various sub-scenarios to evaluate the performance of video quality in the presence of feedback and how user satisfaction varies over time. On the other hand, scenarios are further extended to evaluate cross-layer optimization and non-cross layer optimization and how the multicast impact the goodput, packet delivery delay and quality for scalable video.

Figs. 33(a)-33(b) presents the generic scenarios for unicast and mul-

Table 5: Simulation Parameters

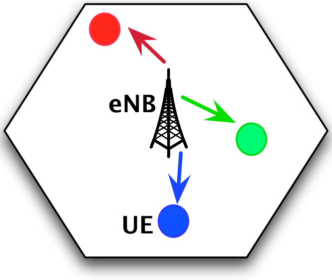
Paramter	Values
T	100
N	10
$\mathcal{L}$	{1-BL, 1-EL} $\rightarrow$ {0,1}
$\mathcal{Z}$	{QPSK, 64-QAM} $\rightarrow$ {0,1}
$\mathcal{S}$	$\{\ \mathcal{S}_{00}, \mathcal{S}_{01}, \mathcal{S}_{10}, \mathcal{S}_{11}\}$
γ	0.6
α	1
β	5
λ	$\{0, \frac{1}{4}, \frac{1}{2}, \frac{3}{4}, 1\}$

ticast video transmissions. Each UE has its own requirement about BL packets and has its own Channel Quality Indicator (CQI) values, therefore, each UE is presented by different color. The content, which eNB transmits to UE is presented by arrow color.

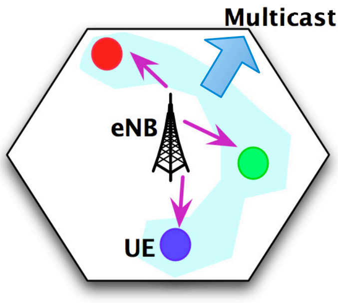
For unicast transmission, as in Fig. 33(a), the eNB transmits what the UE requires based on UE's individual needs, therefore, different contents to different UEs, conversely Fig. 33(b) presents the multicast video transmission from eNB to all the UEs in multicast session where each UE receives same contents, even though they have different requirements. In multicast eNB harmonizes (based on various strategies and policies, as defined in previous chapter) the needs of all UEs and then transmits the same contents to all the UEs.

6.1.1 Feedback Scenario

For feedback scenario, we simulate the system for different values of T to investigate the performance in terms of satisfaction level of the UEs. Each UE has a random loss probability with uniform distribution between 0.01% and 1%. There are four possible combinations of scenarios based on the choice between *Average* and *Maximum* strategies, and including the availability of feedback in multicast environment.



(a) Unicast Scenario



(b) Multicast Scenario

Figure 33: Unicast and Multicast SVC Transmission

Average strategy vs Maximum strategy — In the *Average* strategy, the eNB averages all $Opt_{t_1}(j)$ values required by each UE and then selects the mean value (χ_{avg}) for all the UEs. When feedback from the UE "j" is received after each timeslot about the number of packets received and the perceived quality, the eNB updates its value regarding $Opt_{t_1}(j)$. For the *Maximum* strategy instead, the maximum value amongst $Opt_{t_1}(j)$ for all UEs is selected as χ_{max} for multicast sessions. In fact, thanks to the feedback, the $Opt_{t_1}(j)$ is updated at each transmission and the maximum value is selected again.

Feedback vs Non-feedback cases — In the absence of feedback, the eNB cannot update the $Opt_{t_1}(j)$ at each timeslot; therefore, it will assume the same $Opt_{t_1}(j)$ values and the averaged value χ_{avg} which is initially calculated, if using the *Average* strategy. If the *Maximum* strategy is used, then the eNB selects the maximum $Opt_{t_1}(j)$ amongst all UEs, but does *not* update its value at each transmission because of the absence of feedback. Therefore, even though the strategy is meant to be conservative, due to the chosen value of χ_{max}. some of the users may still be Hopeless or Unhappy because the optimal Opt_{t_1} is determined a priori.

6.1.2 Resource Allocation Scenario

We apply the policies defined in Sec. 5.2.4 to evaluate the performance in the unicast and multicast scenario.

Resource Allocation in Unicast — Based on (5.23) and (5.25), once we compute χ_j, we optimize the allocation of BL and EL packets finding the best MCS assignment for each time slot and for each UE j, individually,

$$\Phi(j)^k \Longrightarrow \begin{cases} \Phi_1^k(\chi_j) \\ \Phi_2^k(t_2(j)) \end{cases} , \tag{6.1}$$

such that

$$\Phi_1^k(\chi_j) = alloc_j^k(\ell, \zeta) \qquad \forall\ (\ell, \zeta) \in \mathcal{L} \times \mathcal{Z} \tag{6.2}$$

$t_2(j) = T - \chi_j$ iff $t_1 \geq \frac{T}{2}$, otherwise $t_2(j) = t_1(j)$. Based on $t_2(j)$, the EL packet allocation is

$$\Phi_2^k(t_2(j)) = alloc_j^k(\ell, \zeta) \iff \zeta > \omega \tag{6.3}$$

where ω (as defined in Sec. 5.2.4) is a threshold dependent on the modulation scheme (in this case QAM) which defines the channel rate required to reliably transmit an EL packet. Thus, if the channel quality is below ω we necessarily transmit a BL packet. The EL packet allocation is similar to the BL packet allocation as follows:

$$alloc_j^k(\ell, \zeta) \Longrightarrow \begin{cases} \arg\max \left(\mathrm{G}_j \mid \mathcal{S}_{\ell\zeta}^k\right) \\ \arg\min \left(\mathrm{D}_j \mid \mathcal{S}_{\ell\zeta}^k\right) \end{cases} , \tag{6.4}$$

Resource Allocation in Multicast Based on the *Average* strategy, the allocation of BL and EL packets for each UE can now be defined as

$$\Phi_{avg}^k \Longrightarrow \begin{cases} \Phi{1}_{j:1\to\mathrm{N}}^k(\chi_{avg}) \\ \Phi{2}_{j:1\to\mathrm{N}}^k(t_2(j)) \end{cases} , \tag{6.5}$$

$$\Phi{1}_{j:1\to\mathrm{N}}^k(\chi_{avg}) = alloc_j^k(\ell, \zeta)\ \forall(\zeta, \ell) \in \mathcal{L} \times \mathcal{Z} \tag{6.6}$$

$$\Phi{2}_{j:1\to\mathrm{N}}^k(t_2(j)) = alloc_j^k(\ell, \zeta) \iff \zeta > \omega \tag{6.7}$$

Here, $alloc_j^k(\ell,\zeta)$ is the same as in (6.4) for both BL and EL packet allocation, Φ_1 and Φ_2, respectively.

Similarly, based on the *Maximum* strategy, the allocation would be

$$\Phi_{max}^k \Longrightarrow \begin{cases} \Phi 1_{j:1\to\mathrm{N}}^k(\chi_{max}) \\ \Phi 2_{j:1\to\mathrm{N}}^k(t_2(j)) \end{cases} , \tag{6.8}$$

$$\Phi 1_{j:1\to\mathrm{N}}^k(\chi_{max}) = alloc_j^k(\ell,\zeta)\ \forall(\ell,\zeta) \in \mathcal{L} \times \mathcal{Z} \tag{6.9}$$

$$\Phi 2_{j:1\to\mathrm{N}}^k(t_2(j)) = alloc_j^k(\ell,\zeta) \iff \zeta > \omega \tag{6.10}$$

6.2 Results

This section discusses the performance evaluation regarding all the aforementioned scenarios. First, analytical results are discussed and then simulation results are presented.

6.2.1 Analytical Results

We take the value of threshold θ_j as a uniformly distributed random variable for each UE j. The analytical results for the optimum number of timeslots Opt_{t_1}, as in (5.14) is presented in Fig. 34, where we plot the Opt_{t_1} versus θ_j. It shows that, as the error probability increases, the UE will need more and more timeslots for BL packets to correctly decode. Another noticing point is that, the higher the threshold of BL packets θ_j, the higher the requirement for Opt_{t_1}.

In Fig. 35 we show the expected quality (5.4) corresponding to the optimal choice of t_1, i.e., Opt_{t_1}, versus p_j. As the loss probability increases, the quality decreases, however, when the β is higher, the quality is also higher. The effect of varying θ_j for single UE while keeping β fixed at 1.5, is presented in Fig. 35. The expected quality is higher when the loss probability neither too high nor too low. Of course, the higher p_j the lower the achieved quality, but this effect seems to be limited, which is

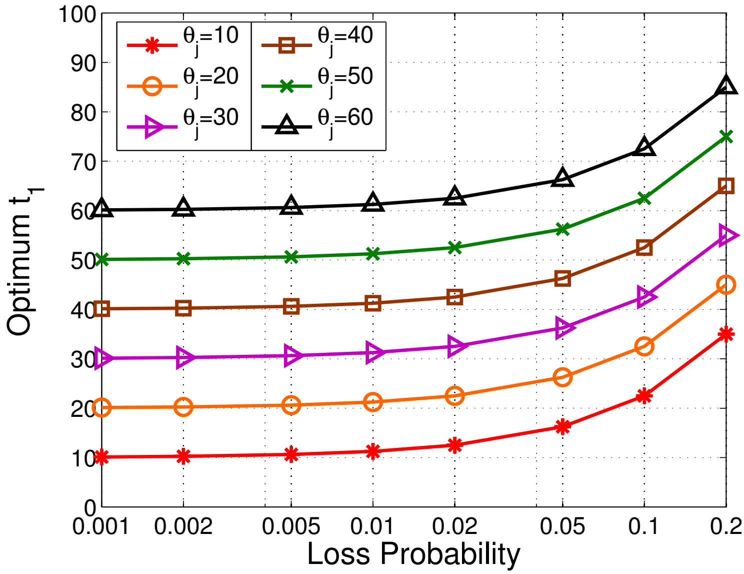


Figure 34: Optimum t_1 With Varying θ_j, $T = 100$ and $\beta = 5$, $\alpha = 1$

likely due to the fact that t_1 is chosen *optimally*, thereby the most satisfactory allocation is achieved for what concern the split between base and enhancement layers. Also note that when $p \ll 1$, then quality brought by a single packet is negligible; the lower the θ_j, the higher the number of EL packets, therefore, the higher the quality received. The impact of the decoding threshold, for which the higher θ_j the lower the quality, seems more relevant. We do not discuss here unicast scenario with respect to feedback and strategy effect, because it will be straightforward.

6.2.2 Simulation Results

The performance of the aforementioned policies is assessed by means of simulation in C++ and Matlab. Each UE receives an H.264/SVC video within T time slots. Based on the channel conditions and on the adopted scheduling policy, the UE will get a certain amount of BL and EL pack-

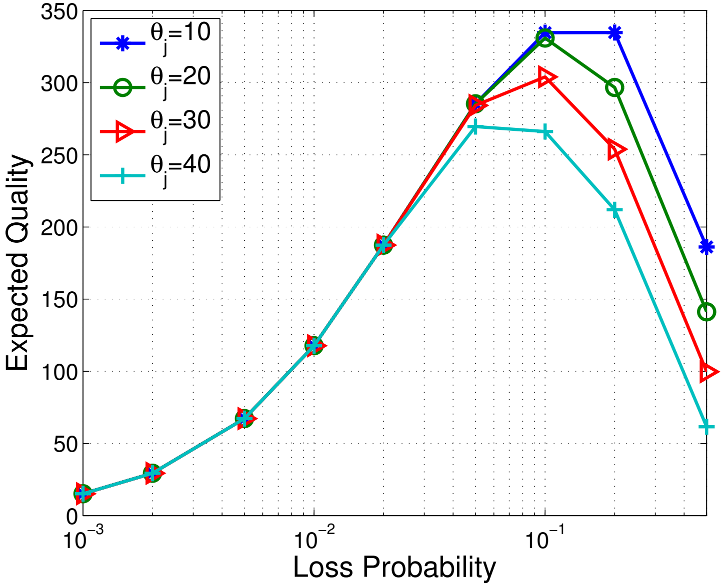


Figure 35: Optimum Quality With Variable θ_j, $T = 100$ and $\beta = 5$

ets. Multiple UEs receive unicast video streams in parallel. This is coordinated by the eNB when the same video is broadcast to all UEs in multicast scenario.

We take into account the unicast and multicast transmissions, and evaluated the performance of allocation policies, results are discussed as follows

Feedback vs Satisfaction

We present the results on the user satisfaction (Sec. 5.2.1) based on the availability of feedback mechanism. Figs. 36-42 represents user satisfaction with various values of T, using *Average* and *Maximum* Strategies.

In Fig. 36, the expected happy users (labeled as "HAP") at the beginning of the transmission are closely related to the actual number of satisfied user at the end of the transmit opportunities. Similarly, the ex-

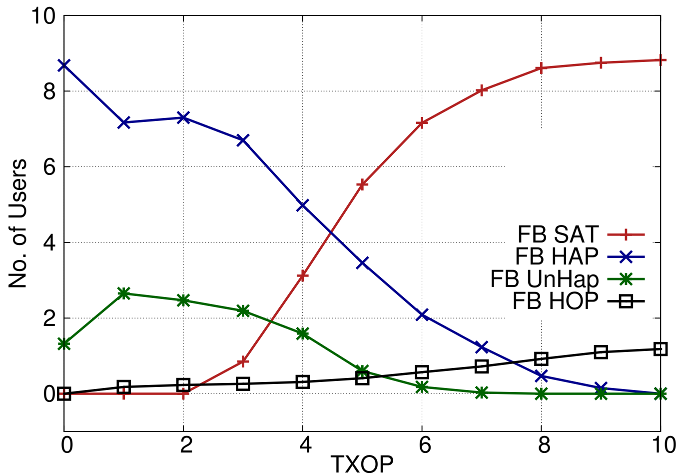


Figure 36: User Satisfaction With Feedback, *Average Policy*, $T = 10$

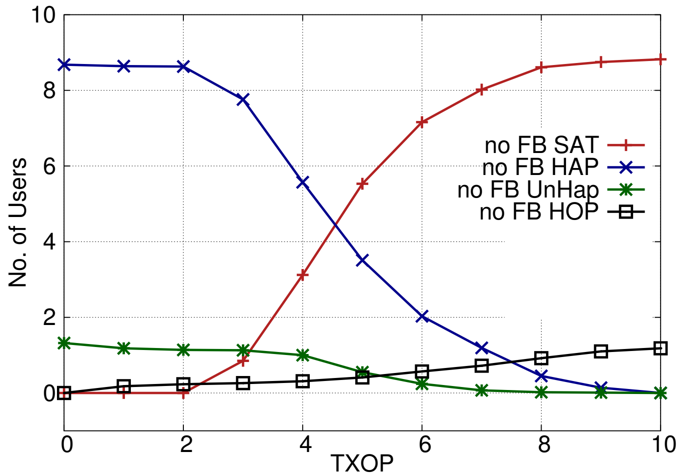


Figure 37: User Satisfaction Without Feedback, *Average Policy*, $T = 10$

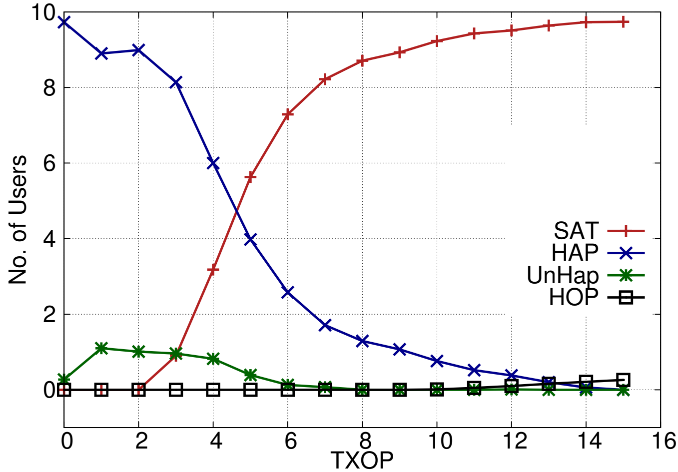


Figure 38: User Satisfaction With Feedback, *Average Policy*, $T = 15$

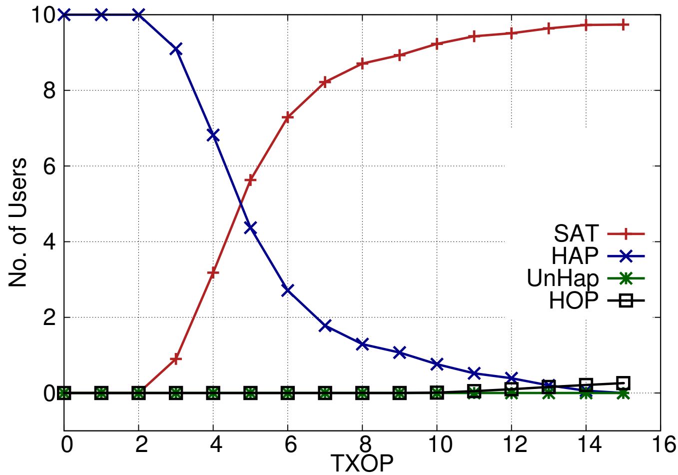


Figure 39: User Satisfaction Without Feedback, *Average Policy*, $T = 15$

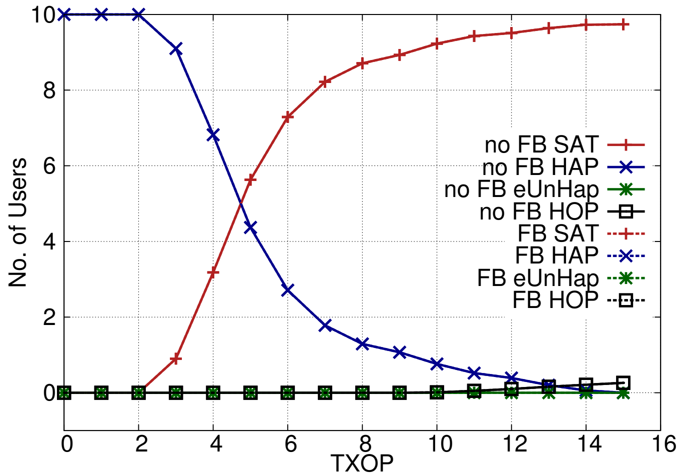


Figure 40: User Satisfaction With Feedback, *Maximum Policy*, $T = 15$

pected unhappy users (labeled as "UnHap") are approximately equal to number of Hopeless (labeled as "HOP") users. However, the curves with feedback and *Average* strategy are more irregular, due to the changes of the strategy when receiving feedback regarding packet losses.

In Fig. 37, the impact of absence of feedback is evaluated while using the same *Average Policy*. It shows that the curves for all satisfaction are not spiky, compared to the result in Fig. 36, where the curves were more noisy due to the feedback and update in policy. Since, there is no feedback in Fig. 37, therefore the eNB does not update the Opt_{t_1} value at each timeslot, and considers the Opt_{t_1} which was computed initially, before transmission.

Similarly, if we consider $T = 15$, the curves get smoother as compare to $T = 10$, case, which means that, for the *Average Policy*, the lower the number of available timeslots the spikier the curves, and vice versa. In Fig. 38, depicts the impact of having feedback while using $T = 15$ with *Average Policy*. Similar to $T = 10$ case, due to the feedback the curves are more spiky, however, noticing point is that, due to higher number of available timeslots, the curves are not much spiky.

On the other hand, for the no-feedback case, with $T = 15$ and *Average Policy*, is illustrated in Fig. 39. It can be seen that the curves are smooth because of the absence of feedback and the eNB relies on a priori information about Opt_{t_1} which was computed before transmission.

The user satisfaction result for *Maximum* strategy is presented in Fig. 40 for $T = 15$. All the users are expected to be initially happy, because the eNB selected the maximum number of transmit opportunities as multicast value. Note that the user satisfaction level is the same regardless of whether the feedback is available or not. Further, the trend of the curves for varying values of T is also the same.

If we compare these four cases, such results are illustrated in Figs. 41 - 42. Here, the dashed curves represent the user satisfaction *with Feedback*, and the solid curves represent the case where there is *no feedback*.

The case for *Average Policy* is presented in Fig. 41, if we compare the feedback versus non-feedback curves, it can be seen that the feedback curves are more spiky, because, the eNB updates the Opt_{t_1} at each timeslot, hence adapt to the current situation. On the other hand, without feedback (solid lines) the curves during transmission are smoother, because the eNB does not update Opt_{t_1} and so, it relies on a priori information about Opt_{t_1} which was computed before the transmission.

As mentioned earlier, the *Maximum Policy*, does not take advantage of feedback. In this policy the eNB tries to satisfy all UEs, hence it computes Opt_{t_1} as maximum value, so that every UE get satisfied. Because, of this policy, the quality of the UEs is effected, which is presented in subsequent sections.

Feedback vs Quality

We present the results for actual quality perceived by UEs, which is averaged over all users for each timeslot and for all simulation runs. Fig. 43 presents the perceived quality, where we compare both *Maximum* and *Average* strategies, as well as checking the difference between presence and absence of feedback.

From Fig. 43, we can see that with the presence of feedback the overall quality is increased compared to the case without feedback. Note that

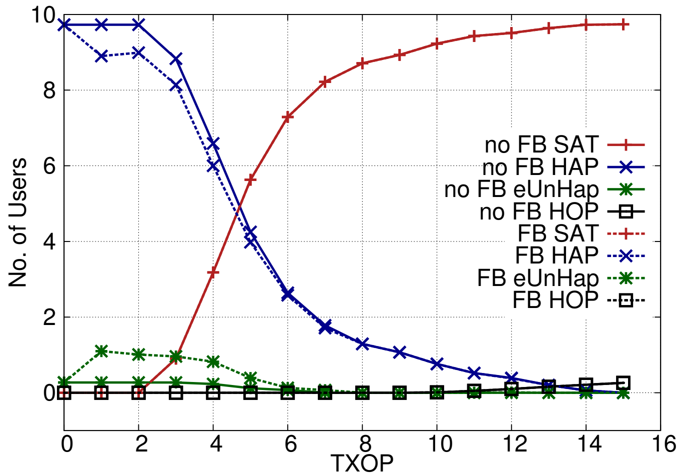


Figure 41: Average Strategy — User Satisfaction — Feedback Comparison — *T=15*

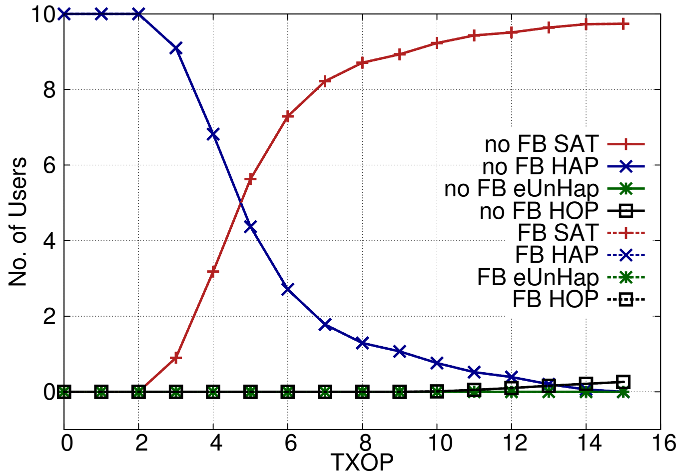


Figure 42: Maximum Strategy — User Satisfaction — Feedback Comparison — *T=15*

by applying the *Maximum* strategy, the quality is eventually lower than by using the *Average* strategy, because the system will transmit BL pack-

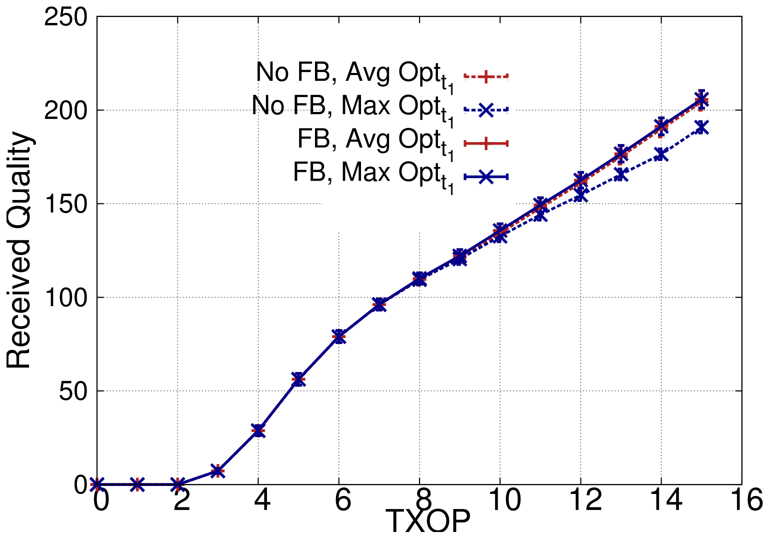


Figure 43: Feedback And Perceived Quality Comparison With $T = 15$, $\beta = 1.5$

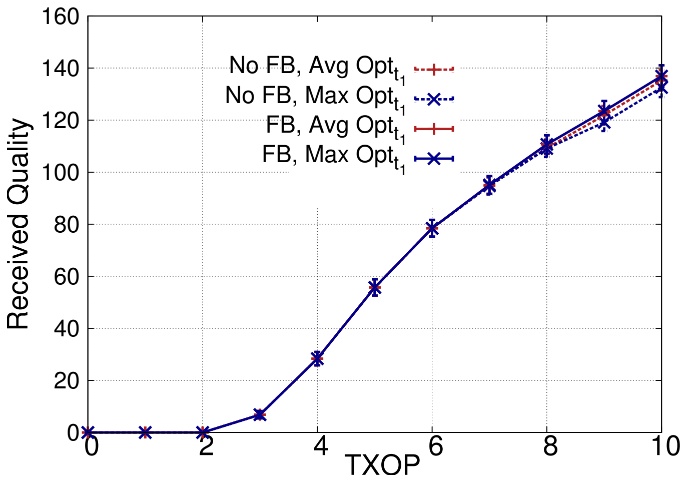


Figure 44: Feedback And Perceived Quality Comparison With $T = 10$, $\beta = 1.5$

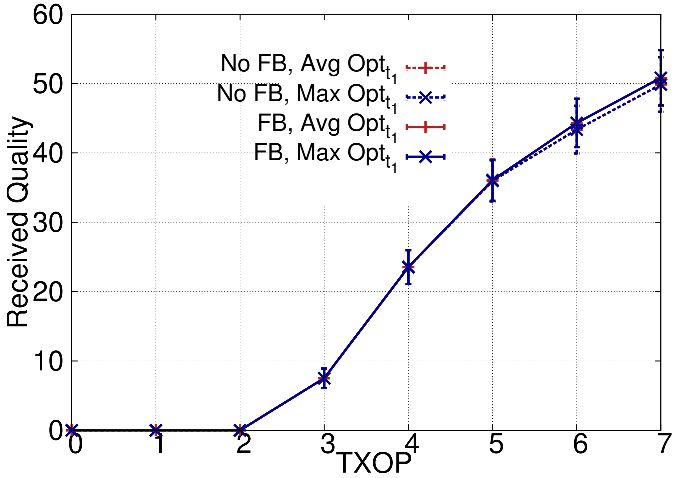


Figure 45: Feedback And Perceived Quality Comparison With $T = 7$, $\beta = 1.5$

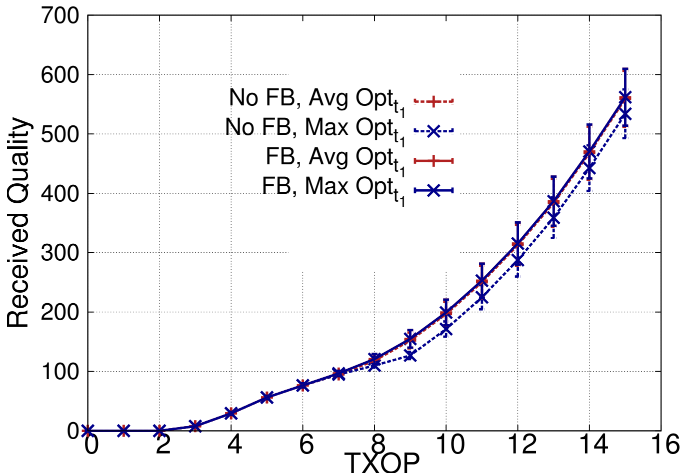


Figure 46: Feedback And Perceived Quality Comparison With $T = 15$, $\beta = 10$

ets for most of the timeslots, and fewer EL packets as compared to the *Average* strategy. Similarly, Fig. 44, show the results for $T = 10$ timeslots.

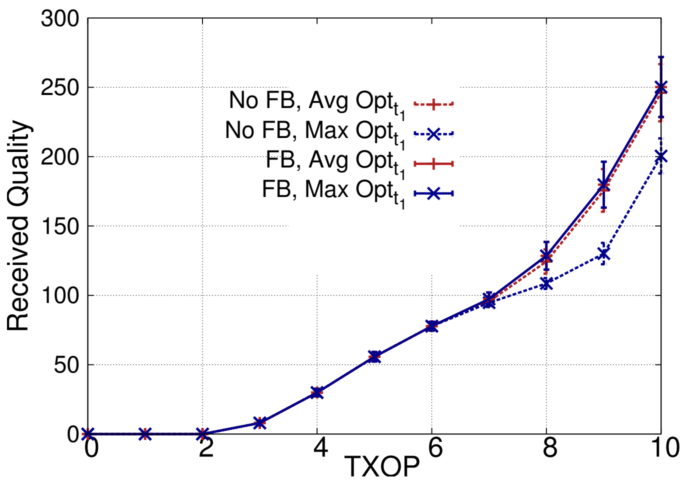


Figure 47: Feedback And Perceived Quality Comparison With $T = 10$, $\beta = 10$

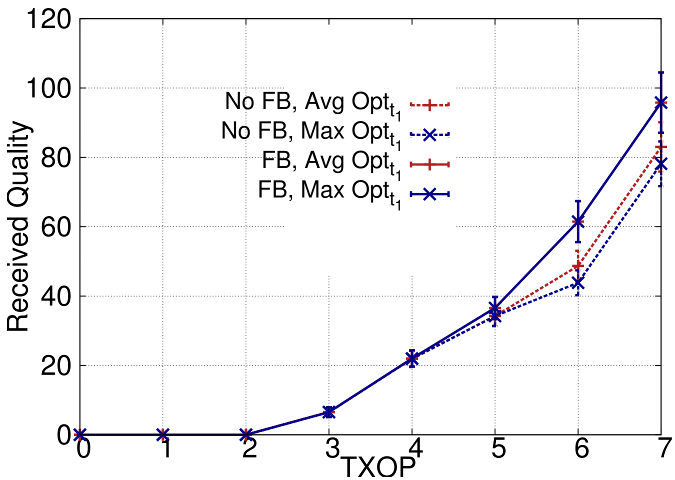


Figure 48: Feedback And Perceived Quality Comparison With $T = 7$, $\beta = 10$

From Figs. 43 and 44, we can see that the lower the total number of timeslots, the smaller the difference between perceived qualities for strategies. However, there is an increase in the standard deviation of all the runs when the number of allowed timeslots are lower. If we increase the β factor to 10, the trend is preserved but the difference between the curves increases as illustrated in Figs. 46 - 47 for $T = 15$ and 10, respectively.

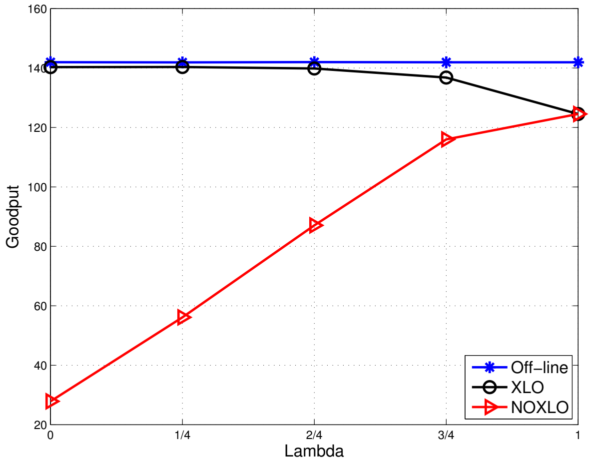


Figure 49: Goodput vs. λ — Unicast Scenarios

Resource Allocation in unicast

In Fig. 49, XLO performs near optimal, i.e., close to the *Offline* solution. The XLO approach takes into consideration the channel condition of the UE and jointly selects the BL/EL packet to be sent and the MCS to use, whereas, the NXLO policy checks the need for BL/EL packets first and then sequentially select the MCS. The performance is reported in terms of goodput, in Fig. 49.

On the other hand, in Fig. 50, XLO has lower packet delivery delay compared to NXLO and near efficient as *Offline*. The quality (5.27) is a combination of goodput and delay, which is defined in Sec. 5.2, which is illustrated in Fig. 51. The curves for quality result lie in between the goodput and delay curves, and the tendency of quality curves can be

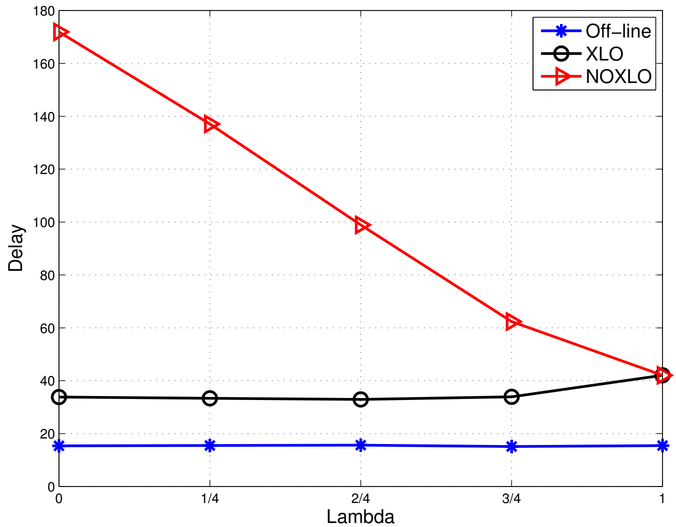


Figure 50: Delay vs. λ — Unicast Scenarios

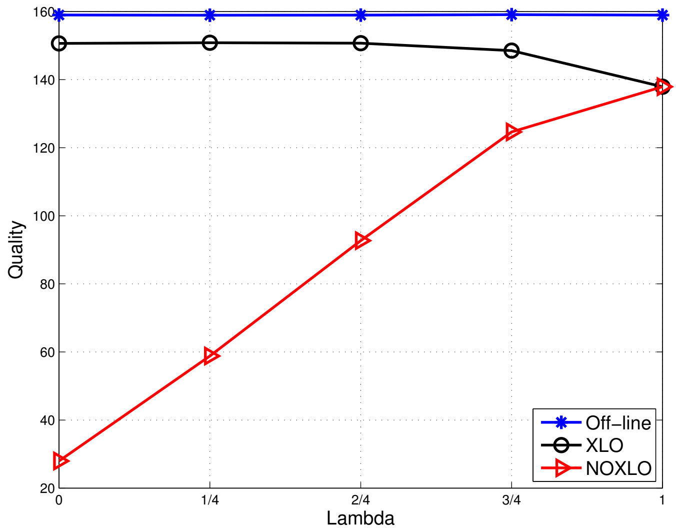


Figure 51: Quality vs. λ — Unicast Scenarios — $\gamma = 0.6$

varied by γ value setting. Here we set $\gamma = 0.6$, to give more wait to goodput, instead of delay.

Resource Allocation in Multicast

The resource allocation for the multicast scenario is based on χ_{avg}, χ_{max}, as calculated in (5.20) and (5.22). In the multicast scenario, the same number of BL packets must be transmitted to each UE. For this reason, the number of BL packets cannot be optimal for all users, but it is the result of a deal among different UEs requirement. Figs. 52-54 show the same trend of the curves in Figs. 49-51, although the performance of policies in multicast is slightly lower, as expected.

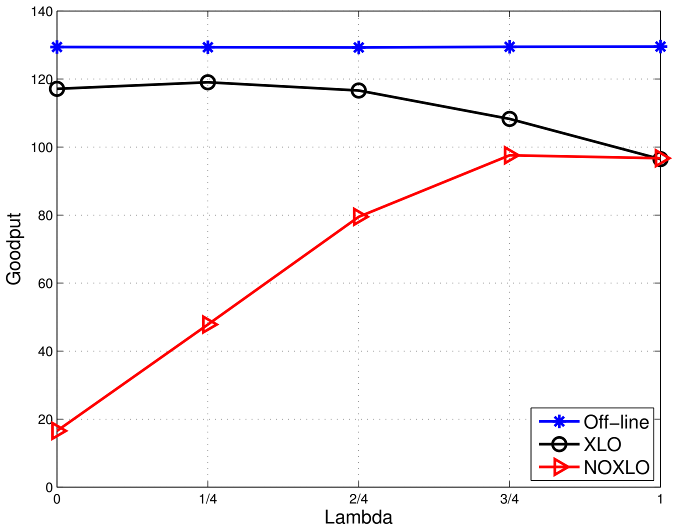


Figure 52: Goodput vs. λ — Multicast Scenarios

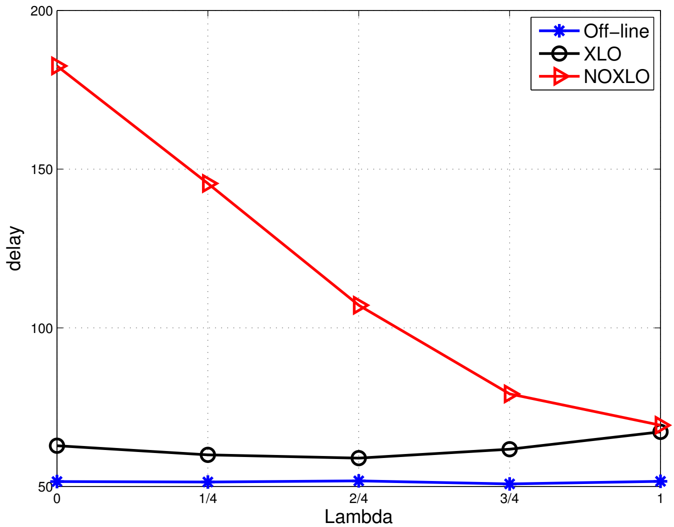


Figure 53: Delay vs. λ — Multicast Scenarios

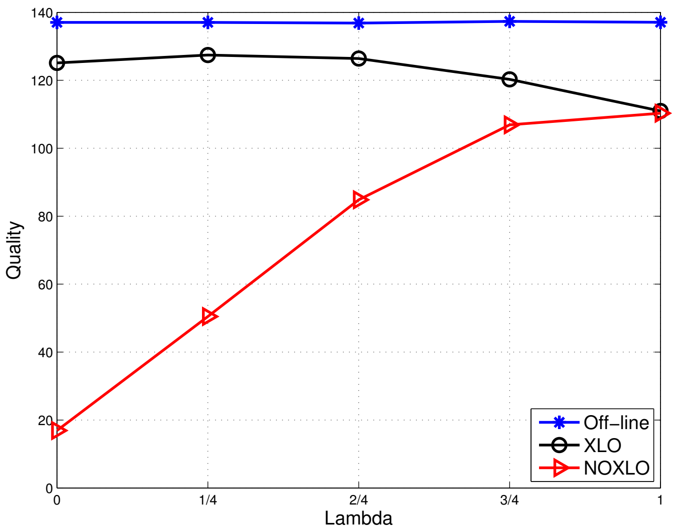


Figure 54: Quality vs. λ — Multicast Scenarios — $\gamma = 0.6$

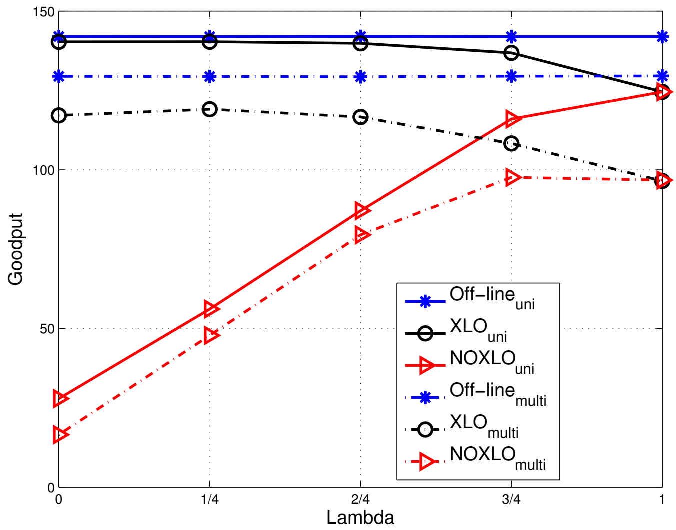


Figure 55: Goodput vs. λ — Unicast/Multicast Comparison

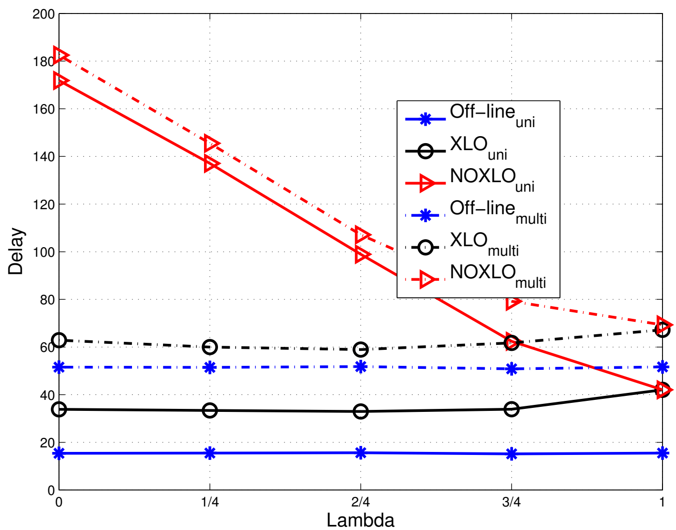


Figure 56: Delay vs. λ — Unicast/Multicast Comparison

6.3 Comparison of Unicast vs Multicast Results

Figs. 55-56 show the results for both unicast and multicast results regarding aforementioned three allocation policies. It can be noticed that the performance of multicast is slightly lower because, all the UEs in a group have to coordinate and might have to compromise for each other.

6.4 QoE Results

To better focus on this comparison, we report more detailed results in Table 6. Here we can see that the unicast performance is better as compared to multicast results, because the eNB considers individual channel condition and BL requirements for unicast whereas, for multicast some users might have to compromise. Another point is that when λ is 1, NXLO and XLO converge and the policy differentiation does not matter for both multicast and unicast results.

Table 6: Comparison Of Unicast And Multicast For Quality

	Offline		**XLO**		**NXLO**	
λ	**Unicast**	**Multicast**	**Unicast**	**Multicast**	**Unicast**	**Multicast**
0	159.0	137.0	150.6	125.1	28.0	16.9
$\frac{1}{4}$	158.9	137.0	150.8	127.4	58.8	50.5
$\frac{1}{2}$	158.9	136.8	150.7	126.4	92.7	84.8
$\frac{3}{4}$	159.1	137.3	148.5	120.3	124.6	106.9
1	159.0	137.1	137.9	111.0	137.9	110.3

6.4.1 Personal Scores Evaluation

We also compute the personal scores of the users about perceived video quality regarding the three different policies.

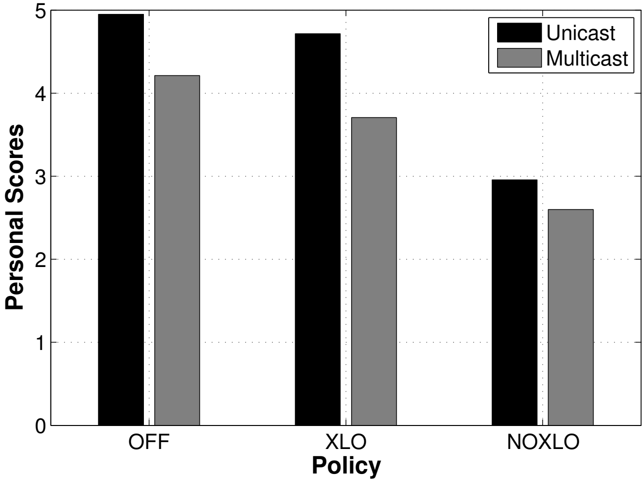


Figure 57: Personal Opinion Scores — Unicast / Multicast Comparison

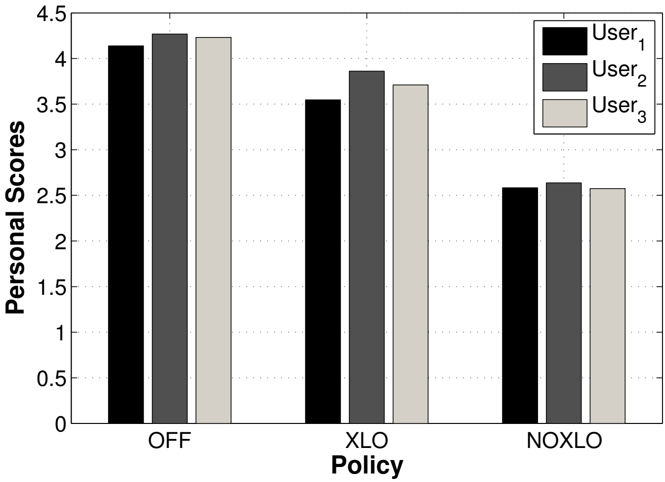


Figure 58: Personal Opinion Scores — User's Comparison

Mean Opinion Score (MOS) has been very well defined and investigated by the [103] regarding the cross-layer optimization for mobile multimedia communications. In this work, the author consider the linear mapping between the MOS and PSNR. We also present the personal scores of the users in terms number of packets received mapping to MOS levels. For our personal score definition, number 5 represent the best maximum one can get, which is the ideal case, whereas, 1 represents the lowest level, which usually *not recommended* level

The results are illustrated in Figs. 57 - 58. It shows that In Fig. 57, the XLO policy score is near optimal (that as *Offline* policy), whereas, for NXLO policy, the score is almost $50 - 60\%$ than the best achievable quality. Another point to be noticed in Fig. 58 is that the UE whose requirement is near averaged value will have more score whereas, the ones who require far less or far more SVC base layer, will have to deal in terms of goodput and delay, and so do for the personal scores as well.

Chapter 7

Conclusions and Future work

7.1 Conclusions

Next Generation Networks are expected to exhibit multimedia contents by 90% as compared to best effort traffic, therefore multimedia quality provisioning is a hot research area nowadays in the networking research community.

Based on the extensive meta-analysis of Chapter-3, the subject taxonomies have been defined focusing on multimedia quality provisioning. Because of the high demands of the multimedia users and extensive use of multimedia applications in the networks it can be seen that the research trend for multimedia quality provisioning is getting hot research topic day by day, and it is expected that it will increase more in future. We believe that the quantitative evaluations performed by this meta-analysis may highlight the existing trend and give guidance to researchers and practitioners, offering a reference value to understand the directions, the scientific community is currently pursuing.

One of our motivation is to study and explore various subject taxonomies for multimedia quality improvement, instead of focusing on only one subject area. That is, the work is done on combination of subject areas as defined in subject taxonomy. Thus, exploiting the layered video contents for enhancing the video QoE by using resource management/optimization techniques in cross-layer fashion for multicast transmission, which is a challenging task. We considered various network management policies in unicast as well as multicast environment for layered video contents.

Main motivation is to investigate the performance of the PHY/ Application cross-layer optimization, for which we consider an analytical model, where the channel is modeled by means of a Markov chain, whose states represent different channel qualities. We handled this challenging research task, by exploiting a Cross-Layer (PHY/ Application) solution with respect to adaptive transmission rates for SVC layers in the multicast environment. We adopt for the Cross-Layer approach by exploiting the application layer information about scalable video layers, that is, base layer and/or enhancement layers, and user channel condition information from the PHY layer for the dynamic assignment of resources at datalink layer by adapting channel rates according to channel condition during the transmission of video contents. This problem gets more crucial when the case of multicast is considered, as the central base station has to consider the heterogeneous requirements and status of all the users, therefore, requiring coordination to harmonize the needs and transmissions of video.

We define the analytical model for optimizing the number of timeslots required to correctly decode the Base Layer for each user by using the Lagrange Multiplier optimization method, where our goal function is to maximize the quality subject to various constraints. To find the maxima, the central base station determine the best split of timeslots for BL and EL packet delivery. The mathematical model considers two packet transmission scenarios. In the former, we consider a single packet transmission per time slot and a single user in the system. This scenario is useful to understand the latter, where multiple packets are transmitted depending on the channel quality, and also multiple users are considered in unicast as well as multicast environment.

We also define the taxonomy of users regarding the satisfaction level of the users during the transmission, and how it changes in the presence of feedback. Each user has its own channel state, packet loss probability, and thus its own requirement about the number of base layer packets needed to decode the video. The strong point of the proposed solution is to give management options for policy selection in the network depending on operator requirements. We also provide a framework for evaluating the importance of feedback in multicast session for layered video contents delivery.

Further, we formulated a model to assess the performance of a Cross-Layer solution as opposed to a sequential selection of the layers. The strong point of the proposed solution is to give management options for policy selection in the network depending on operator requirements. We provide an optimization framework to compute the optimum number of transmit opportunities needed with respect to quality maximization and for evaluating the importance of feedback in multicast session for layered video contents delivery.

For performance evaluations, various scenarios are defined keeping in view various strategies, policies, cross-layer and non-cross-layer optimization techniques, presence of feedback and satisfaction level of users in the unicast as well as multicast environment. Both, the analytical and simulation results are provided for performance evaluation.

In analytical results, it is observed that the higher the requirement of base layer for a user j (that is, θ_j), the higher the number of timeslots required to correctly decoded the video, similarly, if the loss probability is higher, the optimum number of time slots needed for base layer will be higher.

Further, by means of simulation the performance is evaluated in terms of presence of feedback and how user satisfaction level varies during transmission. We also investigated the impact of feedback on the actual perceived quality of the users. We found that, with the presence of feedback, the perceived quality is higher as compare to no-feedback case, whereas, the results for satisfaction levels are more irregular as compare to absence of feedback because of the dynamic nature and updates in policy due to the presence of feedback.

Moreover, resource allocation is evaluated in terms of goodput, packet delivery delay and quality. We observed that the joint selection solution provides near optimal/theoretical best (as performed by *Offline* policy). Finally, we also present QoE results as personal opinion score of the users, where it is shown that user score is high for *Offline* policy being the theoretical bound, but XLO (joint policy) attains quasi-optimal performance, and much better than NXLO, i.e., the sequential policy. Further, in unicast transmission the perceived quality will be higher as opposed to multicast, where transmission is coordinated for all the users, therefore opinion score of the users will be lower in multicast transmission.

7.2 Future Work

An evolution will involve investigating the effect of the number of users in a multicast group and how their joining and leaving can effect the quality of all users.

We plan to apply this study to specific network scenarios, such as the Long Term Evolution of third generation cellular systems, by using specific network simulators.

The idea is to further explore the channel state of users, and utilize efficient resource management techniques in addition to the network policies and feedback mechanisms in such multicast layered video scenario. We plan to extend the analytical model for more channel states and multiple enhancement layers.

How the partial feedback, can improve the results while providing the same level of satisfaction to the end user, instead of exploiting feedback at each timeslot.

As the video technology trend is shifting from 2D to 3D, therefore, the model can be enhanced for 3D video frames, and investigate the QoE improvements.

Another evolution will involve investigating the effect of the number of users in a multicast group and how their joining and leaving can effect the quality of all users. Further, it can also be investigated to compute the PSNR [103] of the video receiver and investigate the effect of feedback on

PSNR and delays.

In video conferencing, particularly for online learning classes, there is a concept of *People+Content* [104] technology, which refers to the multiple video streams (that is, the video/Audio of presenter, powerpoint presentation and whiteboard etc) bundled in one flow. It can be investigated on how to adapt to the end-user requirement for maintaining the satisfaction level of the user while performing efficient resource allocation in terms of fairness to all users, while improving QoE individually.

Appendix A

Appendix

Table 7: Meta Analysis Articles With Respective Fields

Taxonomy	Articles
Packet/ Flow Scheduling	[105], [106], [107], [108], [68], [109], [110], [111], [112], [113], [114], [115], [116], [117], [118], [119], [120], [121], [122], [123], [67], [124], [125], [126], [71], [127], [128], [129], [64], [63], [130], [131], [70], [132], [133], [65], [62], [134], [135], [136], [137], [138], [139], [140], [141], [142], [66], [143], [144]
Resource Management	[145], [144], [146], [147], [148], [149], [150], [151], [152], [153], [154], [155], [156], [157], [158], [159], [37], [160], [161], [162], [163], [164], [165], [166], [167], [168], [169], [170], [171], [34], [172], [173], [174], [175], [78], [176], [177], [45], [178], [54], [179], [180], [181]

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Taxonomy	Articles
Cross Layer design	[46], [47], [182], [183], [39], [40], [184], [185], [186], [187], [188], [189], [190], [191], [192], [193], [194], [35], [38], [195], [196], [197], [105]
Call Admission and Control	[197], [45], [178], [42], [198], [199], [200], [201], [202], [203], [43], [36], [204], [205], [44], [206], [207], [145]
Network Management	[208], [209], [210], [211], [212], [213], [214], [215], [216], [59], [217], [218], [61], [60], [219], [220]
Rate Control	[221], [222], [77], [223], [224], [225], [75], [226], [227], [228], [229], [230], [231], [232], [233], [76], [234], [235], [236], [237], [238], [107]
Mobility Management	[54], [179], [53], [239], [240], [51], [52], [55], [241]
Error Control	[242], [238], [49], [243], [50], [244], [245], [246], [247], [248], [249], [48], [250], [251], [252], [253], [254], [255], [221]
Multicast/ Broadcast Management	[256], [180], [257], [258], [259], [260], [57], [58], [56], [261], [242]
Routing	[262], [263], [264], [265], [266], [267], [80], [268], [269]
Power Control	[255], [241], [181], [269], [270], [72], [73], [256], [106], [222], [77], [223], [146], [147], [148], [262]

Table 8: Meta Analysis Articles With Respective Research Methodologies

Research Methodology	**Articles**
Field Experiment	[253], [254], [55], [208], [215], [220], [67], [128], [270], [230], [267], [244], [217]
Field Experiment	[253], [254], [55], [208], [215], [220], [67], [128], [270], [230], [267], [244], [217]
Laboratory Experimental Research	[189], [193], [252], [52], [59], [125], [136], [142], [236], [171], [261], [121], [143] [255]
Mathematical Analysis	[198], [201], [205], [44], [39], [196], [218], [113], [126], [62], [66], [269], [157], [159], [167], [266], [251], [258], [197], [67], [197], [207], [186], [49], [50], [179], [51], [256], [260], [61], [119], [137], [72], [73], [145], [152], [163]

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Research Methodology	Articles
Simulation	[197], [45], [178], [42], [199], [200], [202], [203], [43], [36], [204], [207], [46], [47], [182], [183], [40], [184], [185], [186], [187], [188], [190], [191], [192], [194], [35], [38], [195], [242], [238], [49], [243], [50], [244], [245], [246], [247], [249], [250], [54], [179], [53], [239], [240], [51], [256], [180], [257], [259], [260], [57], [58], [56], [261], [209], [210], [212], [213], [217], [61], [60], [219], [105], [106], [107], [108], [68], [109], [110], [111], [112], [114], [115], [116], [117], [118], [119], [120], [121], [122], [123], [124], [71], [127], [129], [64], [63], [130], [131], [70], [132], [65], [134], [135], [137], [138], [139], [140], [141], [143], [255], [241], [181], [72], [73], [221], [222], [77], [223], [224], [225], [75], [226], [227], [228], [231], [232], [233], [76], [234], [235], [237], [145], [144], [146], [147], [148], [149], [150], [151], [152], [153], [154], [155], [156], [158], [37], [160], [161], [162], [163], [164], [166], [168], [169], [170], [34], [172], [173], [174], [175], [78], [176], [177], [262], [263], [264], [265], [80], [268], [271], [215], [128], [133], [229], [125], [171], [201]

Table 9: Meta Analysis Articles With Respective Research Models

Research Models	Articles
Listing of variables and implicit relationship	[171], [201], [239], [257], [58], [112], [129], [233], [234], [158], [172]
Listing of variables and levels	[253], [208], [189], [193], [142], [159], [182], [194], [209], [210], [212], [213], [118], [132], [65], [137], [163], [164], [78], [262]
Listing of variables and levels	[253], [208], [189], [193], [142], [159], [182], [194], [209], [210], [212], [213], [118], [132], [65], [137], [163], [164], [78], [262]
Multi-tier Influence Diagram	[254], [215], [270], [267], [214], [216], [229], [52], [59], [125], [196], [178], [202], [43], [46], [40], [184], [188], [238], [243], [246], [249], [53], [51], [180], [260], [61], [105], [106], [117], [120], [121], [131], [221], [223], [228], [231], [232], [235], [149], [150], [151], [155], [37], [160], [162], [173], [265], [268], [208], [67], [144], [168], [54]
Temporal Influence Model	[230], [136], [205], [66], [42], [47], [185], [186], [191], [192], [35], [195], [242], [245], [247], [54], [179], [240], [259], [57], [56], [68], [115], [123], [64], [134], [143], [73], [225], [75], [153], [156], [177], [80], [133], [40]

Table 10: Usage Frequency — Research Methodology vs. Research Model

Research Methodology	Field Experiment	Framework Model	Laboratory Experimental Research	Mathematical Analysis	Simulation
Listing of variables and implicit relationship	0	0	1	1	9
Listing of variables and levels	2	0	3	1	14
Mathematical Model	3	5	2	11	63
Multi-tier Influence Diagram	4	3	3	2	38
No Model	1	1	0	3	8
Temporal Influence Model	1	0	1	2	31

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