Cross Layer Design for Video Streaming over 4G Networks using SVC

by

Rakesh Kappoorkuruppath Valappil Radhakrishna

A thesis submitted to the

Faculty of Graduate and Postdoctoral Studies

In partial fulfillment of the requirements

For the MASc degree in Electrical and Computer Engineering

Ottawa-Carleton Institute for Electrical and Computer Engineering
School of Electrical Engineering and Computer Science
University of Ottawa
Ottawa, Ontario K1N 6N5, Canada

Abstract

Fourth Generation (4G) cellular technology Third Generation Partnership Project (3GPP) Long Term Evolution (LTE) offers high data rate capabilities to mobile users; and, operators are trying to deliver a true mobile broadband experience over LTE networks. Mobile TV and Video on Demand (VoD) are expected to be the main revenue generators in the near future [36] and efficient video streaming over wireless is the key to enabling this. 3GPP recommends the use of H.264 baseline profiles for all video based services in Third Generation (3G) Universal Mobile Telecommunication System (UMTS) networks. However, LTE networks need to support mobile devices with different display resolution requirements like small resolution mobile phones and high resolution laptops. Scalable Video Coding (SVC) is required to achieve this goal. Feasibility study of SVC for LTE is one of the main agenda of 3GPP Release10. SVC enhances H.264 with a set of new profiles and encoding tools that may be used to produce scalable bit streams. Efficient adaptation methods for SVC video transmission over LTE networks are proposed in this thesis. Advantages of SVC over H.264 are analyzed using real time use cases of mobile video streaming. Further, we study the cross layer adaptation and scheduling schemes for delivering SVC video streams most efficiently to the users in LTE networks in unicast and multicast transmissions. We propose SVC based video streaming scheme for unicast and multicast transmissions in the downlink direction, with dynamic adaptations and a scheduling scheme based on channel quality information from users. Simulation results indicate improved video quality for more number of users in the coverage area and efficient spectrum usage with the proposed methods.

Acknowledgements

I would like to acknowledge my sincerest gratitude to my supervisor Prof. Dr. Amiya Nayak, for providing his insights and suggestions that guided me in my thesis research and for providing his dedication, time and encouragement that carried me through the thesis. I thank OPNET Inc. for providing research license to carry out simulations for this thesis.

I am grateful to my friend Balaaji Tirouvengadam for invaluable suggestions and discussions during the thesis work. To my fiancée – Sangeetha, who always understood my busy schedule, I say a heart-felt thanks to you. This thesis would not have been possible without the great support of my family and friends. It is a pleasure to thank them who made this thesis possible.

Finally, this thesis is lovingly dedicated to my parents for their love and support throughout my life.

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Acronyms

1G	First Generation
2G	Second Generation
3G	Third Generation
3GPP	Third Generation Partnership Project
4G	Fourth Generation
AMC	Adaptive Modulation and Coding
AVC	Advanced Video Coding
BL	Base Layer
B frame	Bidirectional predicted frame
BLER	Block Error Rate
BS	Base Station
CDMA	Code Division Multiple Access
CIF	Common Intermediate Format
CQI	Channel Quality Indicator
CSI	Channel State Information
dB	Decibel
EL	Enhancement Layer
eBMSC	evolved Broadcast Multicast Service Center
EDGE	Enhanced Data rates for GSM Evolution
eMBMS	enhanced MBMS
e-NB	evolved NodeB
EPC	Evolved Packet Core
E-UTRAN	Evolved UMTS Terrestrial Radio Access Network
FDD	Frequency Division Duplex
GBR	Guaranteed Bit Rate
GOP	Group of Pictures
GSM	Global System for Mobile Communications
GUI	Graphical User Interface

GW	GateWay
HARQ	Hybrid Automatic Repeat reQuest
HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
I frame	Intra coded frame
IMT	International Mobile Telecommunications
IP	Internet Protocol
ISD	Inter Site Distance
ITU	International Telecommunication Union
ITU – R	International Telecommunication Union – Radio
JSVM	Joint Scalable Video Model
LTE	Long Term Evolution
MAC	Medium Access Control
MANE	Media Aware Network Element
MBMS	Multimedia Broadcast Multicast Service
MBSFN	Multicast Broadcast Single Frequency Network
MCBCS	Multicast Broadcast Service
MCE	Multicast Coordination Entity
MCH	Multicast Channel
MCS	Modulation and Coding Scheme
MIMO	Multiple Input Multiple Output
MMS	Multimedia Messaging Service
MOS	Mean Opinion Score
MPEG	Motion Picture Experts Group
MSP	MCH Scheduling Period
MSAP	MBMS Sub frame Allocation Pattern
NALU	Network Abstraction Layer Unit
OFDMA	Orthogonal Frequency Division Multiple Access
OSI	Open Systems Interconnection
P frame	Predictive coded frame
PHY	Physical Layer

PRB	Physical Resource Block
PSNR	Peak Signal to Noise Ratio
PSS	Packet Switched Streaming
QAM	Quadrature Amplitude Modulation
QCIF	Quarter Common Intermediate Format
QVGA	Quarter Video Graphic Array
QQVGA	Quarter QVGA
QoE	Quality of Experience
QoS	Quality of Service
QP	Quantization Parameter
QPSK	Quadrature Phase Shift Keying
RB	Resource Block
RLC	Radio Link Control
RRM	Radio Resource Management
RTP	Real-time Transport Protocol
SCFDMA	Single Carrier Frequency Division Multiple Access
SE	Spectral Efficiency
SINR	Signal to Interference-Noise Ratio
SNR	Signal to Noise Ratio
SVC	Scalable Video Coding
SS	Subscriber Station
TCP	Transmission Control Protocol
TDD	Time Division Duplex
UDP	User Datagram Protocol
UE	User Equipment
UM	Unacknowledged Mode
UMB	Ultra Mobile Broadband
UMTS	Universal Mobile Telecommunication System
VBR	Variable Bit Rate
VGA	Video Graphic Array
VoD	Video on Demand

WCDMA	Wideband Code Division Multiple Access
WiMAX	Worldwide Interoperability for Microwave Access

Chapter 1

Introduction

1.1 Background

Video streaming over wireless networks is getting popular these days. The main driver for this is the enhanced data rate capabilities of 4G cellular networks. 3GPP LTE technology is the most promising and used 4G technology in the current wireless market. 3G Cellular systems like UMTS use H.264 as video coding standard [52], because of the network friendly bit stream encoding capability and the best coding efficiency available today with H.264. The SVC video coding standard extends H.264 with scalability support. The scalability refers to removal of parts of the bit stream to adapt its capacity to various needs of the end users as well as to varying network conditions. SVC extension of H.264 is a promising approach to deliver scalable content for future mobile video applications. One of the study items in 3GPP Release 10 is about identifying the feasibility of using SVC over LTE networks for improved video support. The SVC encoding process produces a scalable bit stream consisting of a base layer which is backwards compatible to H.264/Advanced Video Coding (AVC), and several enhancement layers. Reception of base layer is enough for sufficient Quality of Experience (QoE). The enhancement layers improve the perceived video quality through higher quality, a higher spatial resolution, a higher frame rate or any combination of these three scalable modes. Overview of SVC and different scalability aspects are explained in [48].

1.2 Motivation

Video streaming over wireless is expected to be one of the main revenue generators for current and future mobile broadband networks [36]. Compared to voice and data services, video streaming over wireless networks is challenging due to the high bandwidth requirement and the delay sensitive nature of video. In the recent past, International Telecommunication Union (ITU) has selected LTE-Advanced and Worldwide Interoperability for Microwave Access (WiMAX) - Advanced as 4G mobile broadband technologies [20]. Quality of Experience similar to fixed broadband networks is expected to be delivered over these mobile technologies. However, the delivery of multimedia services over next generation mobile networks faces some unique challenges when compared to existing wireline networks. Unlike wired networks, in mobile networks the channel quality varies throughout the network. Mobile networks should also support heterogeneous receivers with different processing and display capabilities. Therefore, the video streaming framework for mobile networks should be highly adaptive to mobile device capabilities and network conditions.

H.264/AVC is one of the video codecs defined in 3GPP Release 6 and 3G Multimedia Broadcast Multicast Service (MBMS) recommends only H.264 Baseline profile [1]. However, the SVC extension of H.264 allows efficient temporal, spatial and quality scalabilities [48]. These scalabilities can be used for video bit stream adaptation, based on user and network capabilities. The SVC amendment of the H.264/AVC standard provides network-friendly scalability and adaptability at a bit stream level with a moderate increase in decoder complexity relative to single-layer H.264/AVC. It supports functionalities such as bit rate, frame format, power adaptation and graceful degradation in lossy transmission environments. SVC has achieved significant improvements in coding efficiency with an increased degree of supported scalability relative to the scalable profiles of prior video coding standards. It is also backward compatible with H.264 decoders.

New heterogeneous services emerging in wireless networks require efficient resource management and Quality of Service (QoS) support at all layers of the protocol stack. All the protocol layers should work in better understanding with each other layer for an efficient resource management. As opposed to wireline networks, in which QoS can be guaranteed by independently optimizing each layer in the Open Systems Interconnection (OSI) model, in wireless networks there is a strong interconnection between layers which makes the layered design approach inefficient [27]. Because of this reason, cross layer design is required to optimize the usage of wireless medium and to provide acceptable QoS to wireless users. Cross layer optimized video delivery over LTE is discussed by authors in [28]. Authors are presenting a new scheduling algorithm which uses cross layer information in this paper. Exchange of information between different layers of the transport including Physical (PHY or Layer1), Medium Access Control (MAC or Layer2) and Application layer (Layer 7) of OSI model is used for the proposed algorithm. Scalabilities provided by SVC video and cross layer design provide attractive option for implementing adaptive video streaming algorithms. Especially it is useful for achieving adaptations in MAC layer and application layer using channel quality information from physical layer.

A lot of research has been done on SVC based video transmission over wireless access networks, including LTE. Channel qualities of individual users are available in MAC layer of eNodeB (Base Station in an LTE system, evolved NodeB) in form of Channel Quality Indicator (CQI) feedbacks. Some of the LTE MAC frequency domain scheduling schemes presented in literature uses these feedbacks for channel dependent scheduling. Active dropping of video frames in MAC layer of WiMAX network based on Channel State Information (CSI) is discussed in [19]. Like-wise, Packet adaptation or scheduling in the MAC layer of base station is used by most of the existing works in literature. A cross-layer signaling framework for a dynamic scalable video adaptation in varying network capacity is presented in [40]. The authors compare a fast MAC-layer packet adaptation with a relatively slow and long-term adaptation in the application layer using a real H.264/SVC video. The authors conclude that dropping of packets in video server is a more efficient solution than MAC layer dropping in base station to reduce the congestion in both wired medium in the core network and wireless medium to the UE. Most of the existing literature discusses the adaptations happening in the MAC layer of the base station. But, dropping of packets in the eNodeB is not a common solution for congestion in backhaul routers and wireless medium in a LTE system. Dropped packets in the eNodeB are waste of resources in LTE backhaul and core network. Dropping video frames in the Real Time Transport (RTP) layer of video server reduces the congestion both in eNodeB and backhaul router. Adaptations in video server using channel quality information of LTE networks are not done to the best of our knowledge.

SVC is particularly suitable for multicast because it facilitates the delivery of streaming media to a set of receivers with heterogeneous channel capacities. When a non-scalable video stream needs to be delivered to all users in a multicast group, it has to be streamed at the rate of the weakest user in the group. This significantly limits the utility of users with favorable radio conditions. With the appropriate scheduling algorithms, scalable coding would ensure that the users with good channels receive additional layers and achieve better playback quality. Multicast/broadcast plays a very important role in these entertainment services and applications. Following these demands, 3GPP has defined a Multimedia Broadcast/Multicast Service for UMTS in its Release 6 specification, in which the existing architecture is extended by the introduction of an MBMS bearer service and MBMS user service. In point-to-multipoint mode of MBMS, a group of MBMS subscribers listen to a common channel. They share same time and frequency resources as well as same Modulation and Coding Scheme (MCS). This implies that in order to fulfill QoS requirements, MCS has to be adjusted to the weakest terminal of a subscription group. So, adapting the MCS scheme to the weakest terminal is very important for satisfying the cell edge users. SVC provides an attractive option for sending same video in multiple layers as base layer and enhancement layers. Base layer reception is enough for basic quality of the video, and enhancement layers add up to provide enhanced video quality. This can be used in single-cell MBMS service to provide basic quality to cell edge or low channel quality users and to provide high quality videos to high channel quality users. 3GPP has specified a more advanced and Enhanced MBMS (EMBMS) service, which provides higher frequency efficiency and more reliable point-to-multipoint transmission for LTE. In 3GPP Release 8 specification, the EMBMS transmission is classified into single-cell transmission and Multicast Broadcast Single Frequency Network transmission (MBSFN). In MBSFN operation, MBMS data are transmitted simultaneously over the air from multiple tightly time synchronized cells. These observations lead to the following research problems.

1.2.1 Having efficient video streaming solution for LTE downlink (Problem 1)

Different kind of scalabilities provided by SVC video stream can be used for video bit stream adaptation based on user, radio channel and network capabilities in LTE networks. Advantages of SVC video coding over H.264 for video streaming in LTE networks need to be studied using simulations. Using of scalability features for efficient adaptations based on network and channel quality information also need to be analyzed using real time use cases for an LTE network.

1.2.2 Need for a better adaptation scheme for SVC in unicast (Problem 2)

As mentioned above, most of the existing literature discusses the adaptations happening in the MAC layer of the base station. But, dropping of packets in the eNodeB is not a common solution for congestion in backhaul routers and wireless medium in a LTE system. Dropped packets in the eNodeB are waste of resources in LTE backhaul and core network. Channel quality information from individual users is readily available in MAC layer of eNodeB in the form of CQI feedback. A cross layer method to use this information in the application layer is required to achieve adaptations in the video server itself.

1.2.3 Need for a better adaptation scheme for SVC in multicast (Problem 3)

Some of the previous papers [10, 16] present options of sending one video stream with multiple modulations and coding schemes in wireless networks. But, these schemes are proposed for single cell multicast networks, and MCS selection is not optimized based on user distribution. In MBSFN, cell edge users can add the signals from multiple base stations and get a high quality or medium quality signal. Signal to Interference-Noise Ratio (SINR) values can vary in different regions of cell coverage area depending on the neighboring base station positions. So, adaptation of modulation schemes for different layers need to be done based on channel quality measurements or SINR values of the UEs.

1.3 Objectives and Methodology

The goal of this thesis is to find solutions to the three problems identified in Section 1.2. In particular, the three major objectives are as follows:

i) To compare the advantages of SVC over H.264 video coding method and to develop an efficient video steaming solution for LTE networks using SVC video coding,

ii) To develop channel quality based adaptations in video server using cross layer signaling and SVC video coding,

iii) To develop channel quality based Adaptive Modulation and Coding (AMC) and scheduling scheme for SVC in LTE MBSFN networks

Different kind of information such as network congestion, channel quality, UE power and processing power can be used for achieving cross layer design using SVC video. In this thesis, we will study the adaptations and scheduling in the LTE downlink using channel quality information of individual users. Unicast and multicast video transmission using SVC in LTE networks will be studied with the help of simulations.

We shall use a systematic approach to achieve the above objectives. The first step is to find a simulation tool that could allow the integration of the various part of our model. Of the few available network simulation languages such as Qualnet and NS-2, we shall use OPNET [38] because it supports 3GPP LTE standard and provides an easy to use Graphical User Interface (GUI). It is freely available for research purpose. Moreover, OPNET supports trace based video simulation. Open source MATLAB based LTE system simulator [30] was also considered. But, this simulator only supports full buffer traffic model and not suitable for video traffic simulations. Real video trace files available online [50] and video traces created using Joint Scalable Video Model (JSVM) software [26] shall be used for the simulation of video traffic.

1.4 Contributions

In this thesis, Streaming of SVC encoded videos are studied for improving the perceived video quality by end user with minimum possible throughput consumption in a LTE network. Scalability features of SVC video coding and channel quality information from users are combined to create a cross layer optimized adaptation and scheduling scheme for unicast and multicast video delivery over LTE. LTE MAC layer scheduling and video server adaptation schemes are proposed. Objective video quality metrics and throughput savings are analyzed with and without new algorithms for estimating the benefits of new video stream adaptation algorithms.

The contributions of this thesis are summarized as follows:

i) We analyze video streaming over LTE networks using a 3GPP compliant LTE simulator using H.264 and SVC video traces. Analysis is done using real time use cases of mobile video streaming. Different parameters like throughput, packet loss ratio, delay, and jitter are compared with H.264 single layer video for unicast and multicast scenarios using different kinds of scalabilities. Results show that considerable packet loss reduction and throughput savings (18 to 30%) with acceptable video quality are achieved with proposed scheme based on SVC compared to H.264. This work has been published in IEEE ICPADS 2011 conference [41].

ii) An SVC based video streaming scheme with dynamic adaptations and a scheduling scheme based on channel quality is proposed. Cross layer signaling between MAC and RTP protocols is used to achieve the channel dependent adaptation in video server. An adaptive Guaranteed Bit Rate (GBR) selection scheme based on CQI feedbacks is also presented, and this scheme improves the coverage of the cell. Simulation results indicate improved video quality for more number of users with reduced bit rate video traffic. Approximately 13% video quality gain is observed for users at the cell edge using this adaptation scheme. This work has been accepted for publication [42].

iii) We proposed a video streaming method with SVC for MBSFN networks with AMC and frequency scheduling based on distribution of users in different channel quality regions.

Coverage of low channel quality users is ensured using base layer video sent with low MCS value and higher video qualities are ensured to other users using SVC quality enhancement layers. Through simulations we demonstrate that spectrum savings in the order of 72 to 82% is achievable in different user distribution scenarios with our proposed scheme. This savings in spectrum can be used for serving other MBSFN, single cell MBMS or unicast bearers, and it can also be used for increasing the video quality of the same MBSFN bearer. This work has also been accepted for publication [43].

1.5 Thesis Organization

The rest of the thesis is organized as follows. 4G networks and existing work related to wireless video delivery, scalable video coding and cross layer design in wireless are discussed in Chapter 2. Advantages of SVC over H.264 also are analyzed in chapter 3. Adaptations possible with SVC for video delivery over LTE in unicast and multicast scenarios are also discussed in this chapter. A new bit stream adaptation scheme based on channel quality using cross layer signaling and adaptive GBR selection scheme are presented in Chapter 4. Adaptation and frequency domain scheduling using channel quality information in LTE MBSFN system is discussed in Chapter 5. We give our conclusions in Chapter 6, and discuss the future work.

Chapter 2

Literature Review

Due to the explosive growth of the multimedia Internet applications and dramatic increase in mobile wireless access, there is a significant demand for multimedia services over wireless networks. Furthermore, it is expected that popular content is streamed not just to a single user, but to multiple users attempting to access the same content at the same time. E.g. 3GPP has introduced a new point-to-multipoint optional service named MBMS in Release 6, targeting at simultaneous distribution of multimedia content to many mobile users within a serving area. The expected traffic is believed to be in the areas of weather information, traffic telematics, news broadcast, mobile TV, music streaming, video concert, sports replay, or file sharing. Video transmission in wireless systems is exposed to variable transmission conditions. Furthermore, video content is delivered to a variety of decoding devices with heterogeneous display and computational capabilities. In these heterogeneous environments, flexible adaptation of once-encoded content is desirable. Cisco's recent study on Internet traffic trends [36] projects that by 2013:

- 64% of the mobile traffic will be for video,
- 19% for data services,
- 10% for peer-to-peer, and
- 7% for audio.

The above study results show the importance of video delivery over next generation wireless networks for revenue generation to service providers and operators. Next generation wireless networks need to deliver data capacity and QoE similar to existing wireline solutions, because users are more concerned about QoE and not the technology behind it. So, research is happening all over the world to propose solutions for delivering multimedia data in efficient/cost-effective manner over wireless networks. 3GPP LTE and WiMAX technologies are the strong competitors for 4G of wireless networks. World wide deployment of both of these technologies is happening in a huge scale. Both of these technologies are using Orthogonal Frequency Division Multiple Access (OFDMA) modulation scheme for the downlink.

There is a lot of research happening in the wireless video streaming from different perspective. Currently, H.264/AVC encoder is used for different mobile video services. However, considering the heterogeneous nature of mobile terminal's display and heterogeneous network conditions SVC extension of H.264 is a promising approach to deliver scalable content for future mobile applications.

2.1 Fourth Generation Wireless Technologies

A 4G system is expected to provide a comprehensive and secure all-Internet Protocol (IP) based mobile broadband solution to laptop computer wireless modems, smart phones, and other mobile devices. Facilities such as ultra-broadband Internet access, IP telephony, gaming services, and streamed multimedia may be provided to users. There are a lot of competing technologies for 4G networks. According to ITU-R (International Telecommunication Union – Radio) requirements, an International Mobile Telecommunications (IMT) - Advanced system (4G system) should satisfy following requirements [5].

- Should be based on an all-IP packet switched network.
- Peak data rates of up to approximately 100 Mbit/s for high mobility such as mobile access and up to approximately 1 Gbit/s for low mobility such as nomadic/local wireless access, according to the ITU requirements.
- Dynamically share and use the network resources to support more simultaneous users per cell.
- Scalable channel bandwidth 5–20 MHz, optionally up to 40 MHz.
- Peak link spectral efficiency of 15 bit/s/Hz in the downlink, and 6.75 bit/s/Hz in the uplink (meaning that 1 Gbit/s in the downlink should be possible over less than 67 MHz bandwidth).
- System spectral efficiency of up to 3 bit/s/Hz/cell in the downlink and 2.25 bit/s/Hz/cell for indoor usage.
- Smooth handovers across heterogeneous networks.
- Ability to offer high quality of service for next generation multimedia support.

3GPP LTE, Mobile WiMAX (IEEE 802.16e), Ultra Mobile Broadband (UMB), Flash OFDM and iBurst systems were competing for the 4G status. But, ITU selected LTE-Advanced and WiMAX –Advanced (IEEE 802.16m) proposals for 4G. Both of these technologies have lots of similarities in operation. OFDMA is used as the physical layer access technology in downlink.

Characteristics of various 3GPP technologies are compared in Figure 2.1 and evolution of 4G technologies is shown in Figure 2.2.

Technology Name	Type	Characteristics	Typical Downlink Speed	Typical Uplink Speed
GSM	TDMA	Most widely deployed cellular technology in the world. Provides voice and data service via GPRS/EDGE.		
EDGE	TDMA	Data service for GSM networks. An enhancement to original GSM data service called GPRS.	70 kbps to 135 kbps	70 kbps to 135 kbps
Evolved EDGE	TDMA	Advanced version of EDGE that can double and eventually quadruple throughput rates, halve latency and increase spectral efficiency.	175 kbps to 350 kbps expected (Single Carrier) 350 kbps to 700 kbps expected (Dual Carrier)	150 kbps to 300 kbps expected
UMTS	CDMA	3G technology providing voice and data capabilities. Current deployments implement HSPA for data service.	200 to 300 kbps	200 to 300 kbps
HSPA	CDMA	Data service for UMTS networks. An enhancement to original UMTS data service.	1 Mbps to 4 Mbps	500 kbps to 2 Mbps
HSPA+	CDMA	Evolution of HSPA in various stages to increase throughput and capacity and to lower latency.	1.9 to Mbps to 8.8 Mbps in 5/5 MHz Approximate doubling with dual carrier in 10/5 MHz	1 Mbps to 4 Mbps in 5/5 MHz or in 10/5 MHz
LTE	OFDMA	New radio interface that can use wide radio channels and deliver extremely high throughput rates. All communications handled in IP domain.	6.5 to 26.3 Mbps in 10/10 MHz	6.0 to 13.0 Mbps in 10/10 MHz
LTE-Advanced	OFDMA	Advanced version of LTE designed to meet IMT-Advanced requirements.		

Figure 2.1: Characteristics of 3GPP Technologies [32]

Since, we are going to study about downlink video streaming in LTE, a brief description of LTE and OFDMA is given below in Section 2.2.

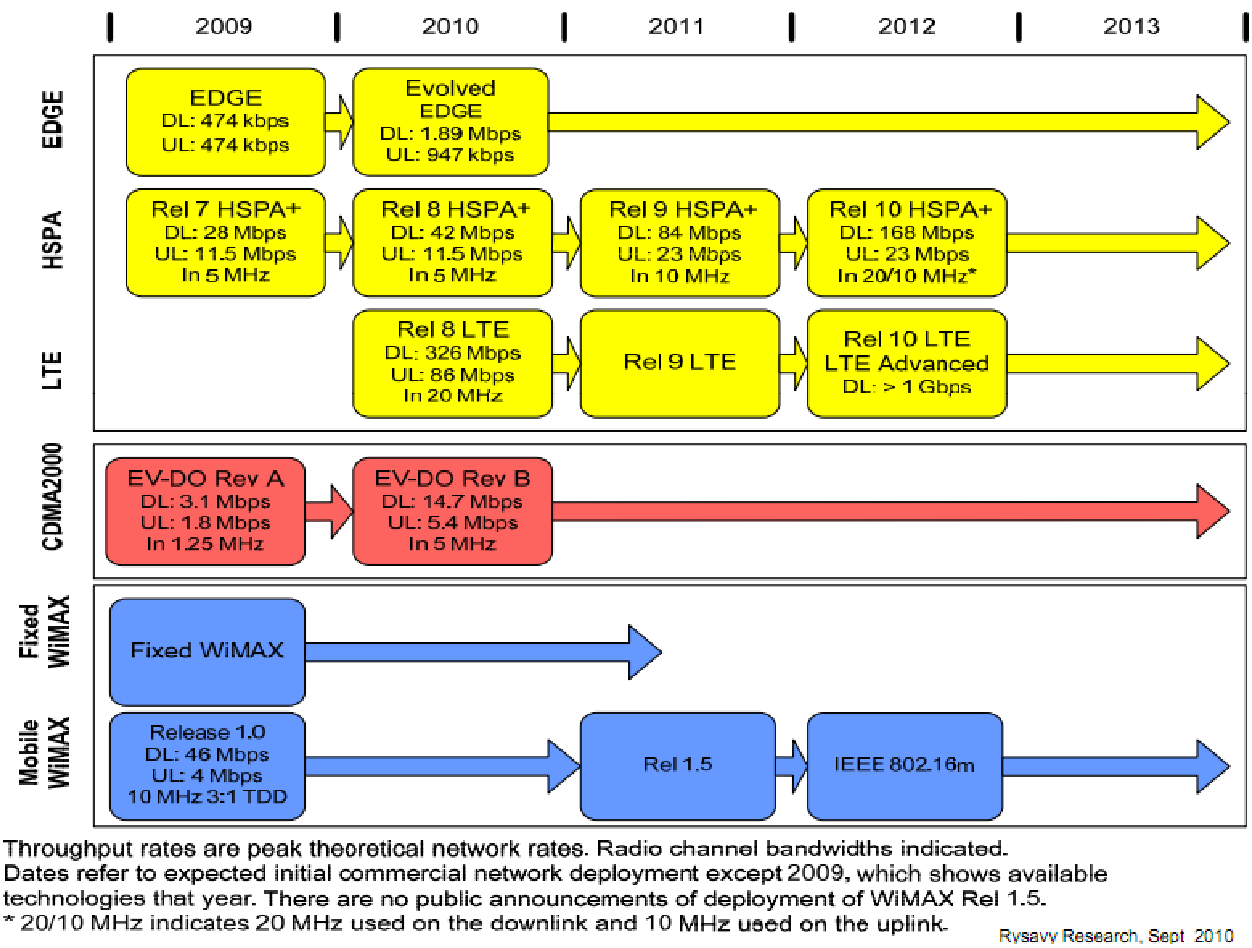

Figure 2.2: Evolution of IMT-Advanced Standard (4G) [33]

2.2 LTE and OFDMA

3GPP LTE system is proposed as a new radio access technology in order to support high-speed data and media traffic. LTE improves the UMTS mobile phone standard and provides a greatly enhanced user experience for the next generation mobile broadband. LTE network is comprised of the access network and the core network, known as Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC), respectively. The LTE system supports both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) schemes. Different carrier frequencies ranging from 1.4 MHz to 20 MHz are supported. The 3GPP LTE system aims to provide high throughput, high spectrum efficiency as well as low latency. The 3GPP LTE system architecture consists of only eNodeBs between the users and the core network. All radio resource management (RRM) functions are performed at eNodeBs. Packet scheduling is one of the RRM mechanisms and the 3GPP LTE system allocates the available system radio resources to active users based on the packet scheduling algorithms.

The evolved UMTS terrestrial radio access (E-UTRA) system of LTE uses OFDMA for the downlink and single-carrier FDMA (SCFDMA) for the uplink. The downlink transmission scheme of FDD and TDD are based on conventional orthogonal frequency division multiplexing (OFDM). OFDMA is a multi-user version of the popular OFDM digital modulation scheme. The available spectrum is divided into multiple resource blocks based on the time and frequency domains. A resource block is the smallest allocation unit in LTE OFDM radio resource scheduling, which can be independently modulated by a low-rate data stream. As compared to OFDM, OFDMA allows multiple users to access the available bandwidth and assigns specific time-frequency resources to each user; thus, the data channels are shared by multiple users. Based on feedback information about the channel conditions, adaptive user-to-subcarrier assignment can be achieved. If the assignment is done sufficiently fast, this further improves the OFDM robustness to fast fading and narrow-band co-channel interference, and makes it possible to achieve even better system spectral efficiency. Different number of sub-carriers can be assigned to different users, in view to support differentiated QoS, i.e. to control the data rate and error probability individually for each user. The OFDMA technology allows high performance in frequency selective channels. In every scheduling interval, the resource that is allocated to a user in the downlink is called a resource block (RB). In the frequency domain, the RB consists of 12 sub-carriers (total bandwidth of 180 kHz) and, in the time domain it is made up of one time slot of 0.5 ms duration. The subcarriers in LTE have a constant spacing of 15 kHz. The resource block size is the same for all bandwidths. The Physical Resource Block (PRB) has both frequency and time aspects: it consists of 12 subcarriers in frequency and 2 consecutive time slots each of which is 0.5 ms long in time. Structure of PRBs in LTE bandwidth and an example allocation pattern of PRBs in OFDMA frame are shown in Figure 2.3.

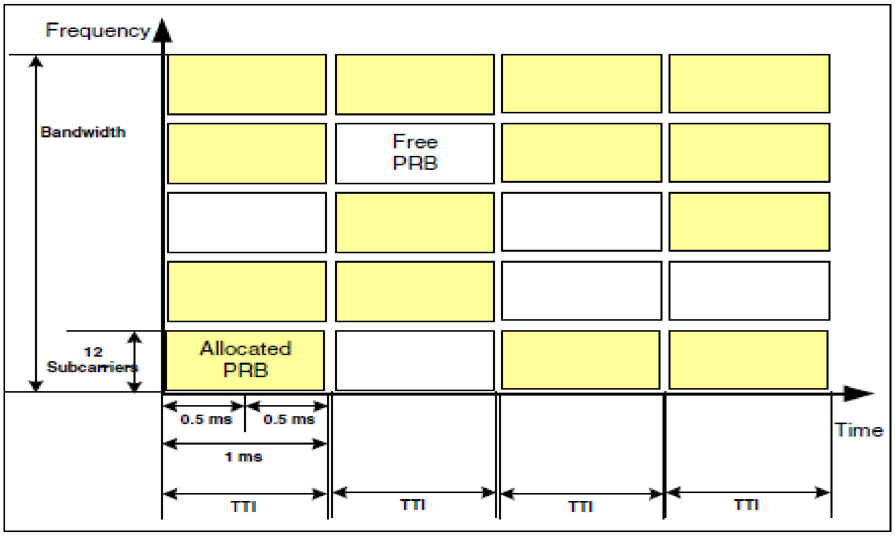

Figure 2.3: Structure and allocation of PRBs in OFDMA [8]

2.2.1 Video Delivery over LTE

Figure 2.4 illustrates the overall architecture of real-time video delivery over LTE cellular networks, in which the network is comprised of the access network and the core network, known as Evolved UTRAN and EPC respectively. Revenue generation and the success of next generation mobile operators depend upon value added services like mobile TV and VoD. There are many driving factors for improved video support in LTE networks. However, the most important one is the high throughput possible with LTE technology. This high data rate is possibly due to following system design decisions.

Scalable channel bandwidth: LTE supports bandwidths from 1.4 MHz to 20 MHz.

Dynamic modulation: A wide range of modulation schemes from Quadrature Phase Shift Keying (QPSK) to 64 Quadrature Amplitude Modulation (QAM) modulations are possible in LTE downlink.

Multiple antenna technology: LTE supports up to 4X4 Multiple Input Multiple Output (MIMO) antenna configurations.

OFDMA: OFDMA in LTE downlink allows radio resources to be allocated in time and frequency domain. This gives link and channel aware schedulers more flexibility for the efficient use of radio resources.

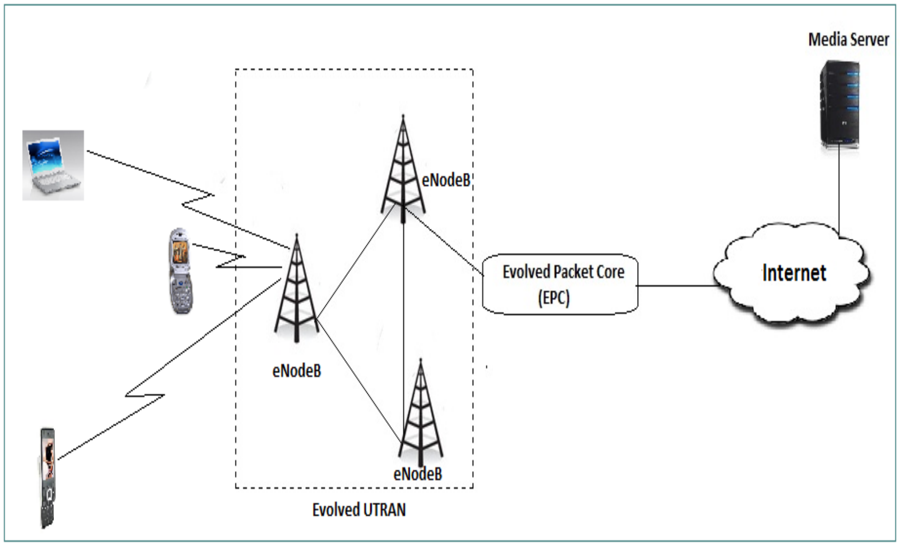

Figure 2.4: Real time video delivery over LTE

Audio and video media coding standards for LTE are discussed in [22]. 3GPP recommends the use of H.264/AVC baseline profiles for all video based services, like conversational, packet switched streaming (PSS) services, Multimedia Messaging Service (MMS) and MBMS services [48]. H.264/AVC baseline profiles are used for delivering video streaming services in 3G networks. But, LTE networks need to support mobile devices with different display resolution requirements like small resolution mobile phones and high resolution laptops. Recognizing this trend, 3GPP research has started focusing the use cases for advanced video support in 4G networks and is evaluating different video codecs. The goal of this work is to further the support of advanced terminals with higher device capabilities in a bandwidth efficient manner. As a result of the 3GPP Release 9 work, a technical report that collects the identified use cases and solutions has been produced [3]. These solutions are based on SVC encoding of video. The evaluation of scalable video coding for LTE is one of the main agenda of 3GPP Release10. SVC has been defined as an extension to the H.264 video coding standard. SVC enhances H.264 with a set of new profiles and encoding tools that may be used to produce scalable bit streams. SVC supports mainly three different types of scalability: spatial scalability, temporal scalability and, quality scalability. Using temporal scalability, the same video sequence can be encoded to support different frame rates. Spatial scalability, on the other hand, is the most important scalability type in SVC. Using spatial scalability, same video can be encoded in multiple resolutions and send using only one bit stream. Spatial scalability is an important tool in wireless networks that support heterogeneous User Equipment (UE) types. Finally, quality scalability produces video bit streams with different quality layers, which can be used for the dynamic adaptation of video quality and bit rate based on user and network conditions. SVC allows different types of scalabilities to be encoded in the same bit stream.

2.2.2 Channel Quality in LTE

In the LTE, subcarriers are assigned to users in the chunks called physical resource blocks. Upon receiving a downlink signal, the UE should report its Signal to Noise Ratio (SNR) to eNodeB via upload control channel. These reports are called CQI reports. CQI values are evaluated from SNR measurements in the UE. CQI value indicates the highest modulation and the code rate at which the block error rate (BLER) of the channel being analyzed does not exceed 10 %. Compared to 3GPP UMTS systems, an advanced CQI feedback system is implemented for 3GPP LTE. Sub-band and wideband CQI reports can be obtained from the UE when CQI reporting scheme is enabled. Sub-band CQIs give channel state information for each sub-band, and wideband CQI gives average channel quality information for the entire spectrum. Sub-band CQIs are used for link adaptation purpose and channel dependent scheduling. Concept of wideband CQI, sub-band CQI and different CQI reporting modes are explained in LTE physical layer standard [2]. There are two CQI reporting modes used in LTE.

- aperiodic feedback: UE sends CQI only when it is asked to by the Base Station (BS).
- periodic feedback: UE sends CQI periodically to the BS; the period between 2 consecutive CQI reports is communicated by the BS to the UE at the start of the CQI reporting process.

An approximate mapping between SNR and CQI is given in [21] and shown below in Figure 2.5.

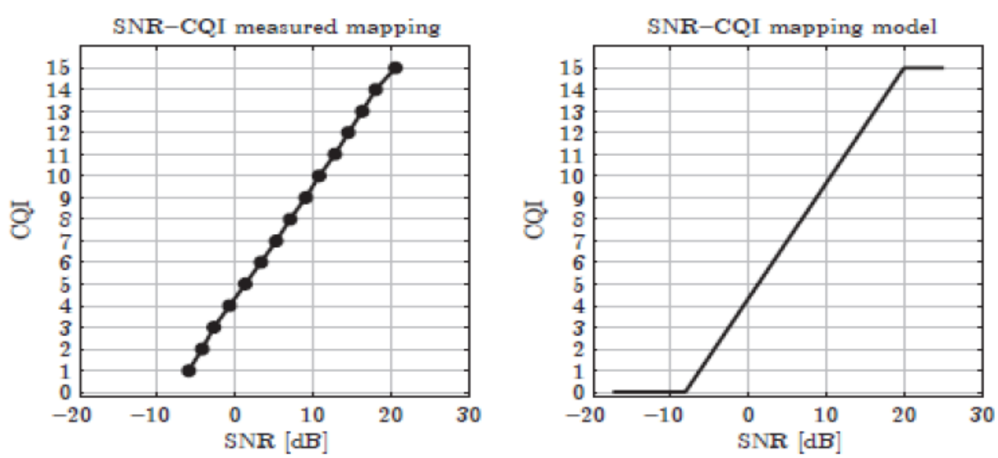

Figure 2.5: SNR to CQI mapping [21]

2.2.3 MBSFN in LTE

3GPP has specified a more advanced and EMBMS service, which provides higher frequency efficiency and more reliable point-to-multipoint transmission for LTE. In 3GPP Release 8 specification, the EMBMS transmission is classified into single-cell transmission and MBSFN. In MBSFN operation, MBMS data are transmitted simultaneously over the air from multiple tightly time synchronized cells. A group of those cells which are targeted to receive these data constitute the so-called MBSFN area [4]. Since the MBSFN transmission greatly enhances the Signal to Interference Noise Ratio (SINR), MBSFN transmission mode leads to significant improvement in spectral efficiency (SE) compared to single-cell MBMS service. This is extremely beneficial at the cell edge, where transmissions from neighboring cells (which are considered as interference in single-cell MBMS) are combined into useful signal energy, and hence the received signal strength is increased while, at the same time, the interference power is largely reduced. Within E-UTRAN, the evolved Node Bs or base stations (e-NBs) are the collectors of the information that has to be transmitted to users over the air-interface. The Multi-cell/multicast Coordination Entity (MCE) coordinates the transmission of synchronized signals from different cells (e-NBs). MCE is responsible for the allocation of the same radio resources, used by all e-NBs in the MBSFN area for multi-cell MBMS transmissions. Besides allocation of the time/ frequency radio resources, MCE is also responsible for the radio configuration, e.g. the selection of the MCS. The e-MBMS Gateway (e-MBMS GW) is physically located between the evolved Broadcast Multicast Service Center (e-BMSC) and e-NBs, and its principal functionality is to forward the MBMS packets to each e-NB transmitting the service. The e-MBMS GW is logically split into two domains. The first one is related to control plane while the other one is related to user plane. Likewise, two distinct interfaces have been defined between e-MBMS GW and E-UTRAN, namely M1 for user plane and M3 for control plane. M1 interface makes use of IP multicast protocol for the delivery of packets to e-NBs. M3 interface supports the MBMS session control signaling, e.g. for session initiation and termination. The e-BMSC is the entity in charge of introducing multimedia content into the 4G network. For this purpose, the e-BMSC serves as an entry point for content providers or any other broadcast/multicast source which is external to the network [4]. The LTE MBSFN system architecture is shown in Figure 2.6.

Scheduling of Multicast Transport Channel (MCH) is done in MCE, whereas unicast scheduling is done in eNodeB. Sub frames used for a MBSFN bearer is changed dynamically in each MCH Scheduling Period (MSP). In MBSFN, the MCE allocates radio resources to the MCH, by the means of an MSAP (MBMS Sub frame Allocation Pattern). Channel estimation of MBSFN users is done using MBSFN reference signals instead of cell-specific reference signals in unicast transmission. Since aggregated channel of all cells involved in the MBSFN transmission will be highly frequency selective, MBSFN reference signals are repeated more frequently compared to cell-specific reference signals.

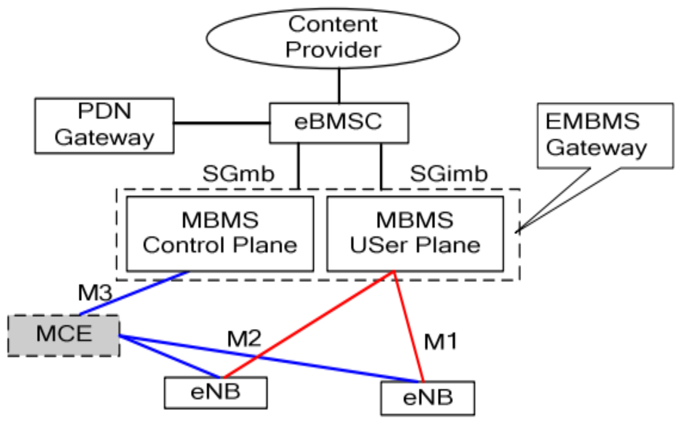

Figure 2.6: LTE MBSFN system architecture [54]

Authors of [12] present a packet scheduler for UMTS MBMS services. This packet scheduler is designed to deliver two different quality of service for users located in different locations of the same sector of a cell and power requirement of MBMS service is reduced with this. Some of the previous papers [10, 16] present options of sending one video stream with multiple modulations and coding schemes in cellular networks. But, authors use multicasting in single cell networks and MCS decisions are not optimized based on user distribution.

SE refers to the information rate that can be transmitted over a given bandwidth in a specific communication system. It constitutes a measure of how efficiently a limited frequency spectrum is utilized. This can be increased by using high modulation and coding index. Authors of [6] compare SE of MBSFN based LTE networks using different MCS selection algorithms. A typical unicast transmission will have 14 OFDM symbols per sub frame. However, only 12 OFDM symbols per sub frame are used for MBSFN and only one sub frame per radio frame is allowed to use for one MBSFN bearer. Moreover, MBSFN reference signals are transmitted more frequently compared to cell-specific reference signals used for unicast transmission. These factors reduce the SE compared to unicast and total cell MBSFN SE calculations.

2.3 H.264 and Scalable Video Coding

International video coding standards such as H.261, MPEG-1, H.262/MPEG-2 Video, H.263, MPEG-4 Visual, and H.264/AVC have played an important role in the success of digital video applications. The H.264/AVC specification represents the current state-of-the-art in video coding. Compared to prior video coding standards, it significantly reduces the bit rate necessary to represent a given level of perceptual quality – a property also referred to as increase of the coding efficiency. For 3GPP MBMS video services, H.264/AVC baseline profile is the recommended video codec. Scalability has already been present in these video coding standards in the form of scalable profiles. However, the provision of spatial and quality scalability in these standards comes along with a considerable growth in decoder complexity and a significant reduction in coding efficiency (i.e., bit rate increase for a given level a reconstruction quality) as compared to the corresponding non-scalable profiles. These drawbacks, which reduced the success of the scalable profiles of the former specifications, are addressed by the new SVC amendment of the H.264/AVC standard.

SVC standard provides network-friendly scalability and adaptability at a bit stream level with a moderate increase in decoder complexity relative to single-layer H.264/AVC. It supports functionalities such as bit rate, format, and power adaptation, graceful degradation in lossy transmission environments. It is backward compatible with H.264 decoders. SVC has achieved significant improvements in coding efficiency with an increased degree of supported scalability relative to the scalable profiles of prior video coding standards. The desire for scalable video coding, which allows on-the-fly adaptation to certain application requirements such as display and processing capabilities of target devices, and varying transmission conditions, originates from the continuous evolution of receiving devices and the increasing usage of transmission systems that are characterized by a widely varying connection quality. A typical video streaming environment with heterogeneous receivers is shown in Figure 2.7.

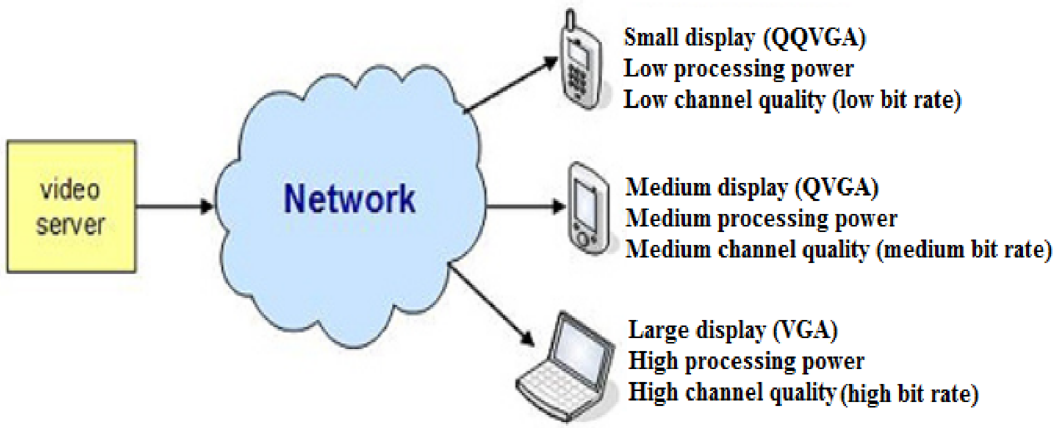

Figure 2.7: Streaming with heterogeneous receiving devices and varying network conditions [45]

Scalable video coding and its advantages for wireless video streaming are briefly outlined in references [48] and [49]. Effect of packet losses on video quality is very important for further study on scalability and adaptability of video traffic. It is analyzed using mathematical models in [15]. A detailed study on system and cross layer design for mobile video transmission is presented in [46]. SVC standard is discussed in reference [45].The objective quality assessment of multidimensional scalability of SVC videos is discussed in [13]. According to the authors, conventional objective quality assessment tools like Peak-Signal-to Noise Ratio (PSNR) fail for scalable video.

A video bit stream is called scalable when parts of the stream can be removed in a way that the resulting sub-stream forms another valid bit stream for some target decoder, and the sub-stream represents the source content with a reconstruction quality that is less than that of the complete original bit stream but is high when considering the lower quantity of remaining data. Bit streams that do not provide this property are referred to as single-layer bit streams. The usual modes of scalability are temporal, spatial, and quality scalability. Spatial scalability and temporal scalability describe cases in which subsets of the bit stream represent the source content with a reduced picture size (spatial resolution) or frame rate (temporal resolution), respectively. With quality scalability, the sub-stream provides the same spatio-temporal resolution as the complete bit stream, but with a lower fidelity – where fidelity is often informally referred to as SNR. Quality scalability is also commonly referred to as fidelity or SNR scalability. The different types of scalability can also be combined, based on the application and network requirements. One SVC-based video stream consists of a base layer (BL) and several enhancement layers

(ELs). Reception of base layer is enough for basic quality video and reception of enhancement layers improve the video quality, frame rate or spatial resolution based on the scalability used. Concept of scalabilities in SVC is shown in Figure 2.8.

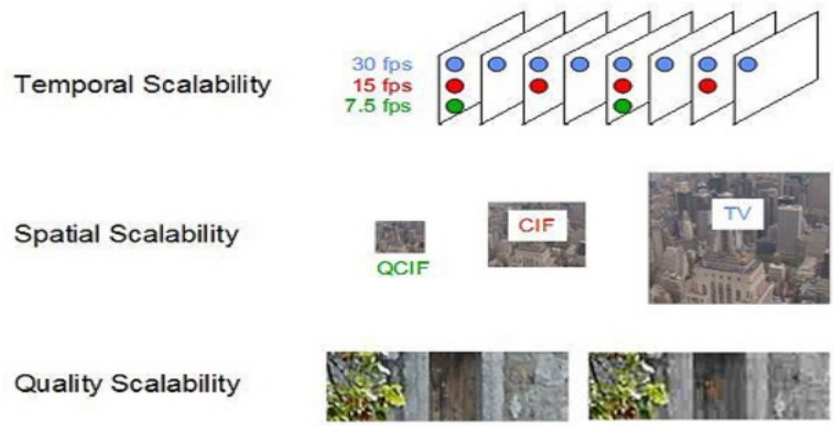

Figure 2.8: The basic types of scalability in video coding [45]

A. Temporal Scalability

A bit stream provides temporal scalability when the set of corresponding access units can be partitioned into a temporal base layer and one or more temporal enhancement layers with the following property. Let the temporal layers be identified by a temporal layer identifier *T*, which starts from 0 for the base layer and is increased by 1 from one temporal layer to the next. Then for each natural number *k*, the bit stream that is obtained by removing all access units of all temporal layers with a temporal layer identifier *T* greater than *k* forms another valid bit stream for the given decoder.

Temporal scalable bit-stream can be generated by using hierarchical prediction structure without any changes to H.264/MPEG-4 AVC. Key pictures are coded in regular intervals by using only previous key pictures as references. The pictures between two key pictures are hierarchically predicted as shown in Figure 2.9. Different temporal layers *T0, T1, T2* and *T3* are shown in Figure 2.10.

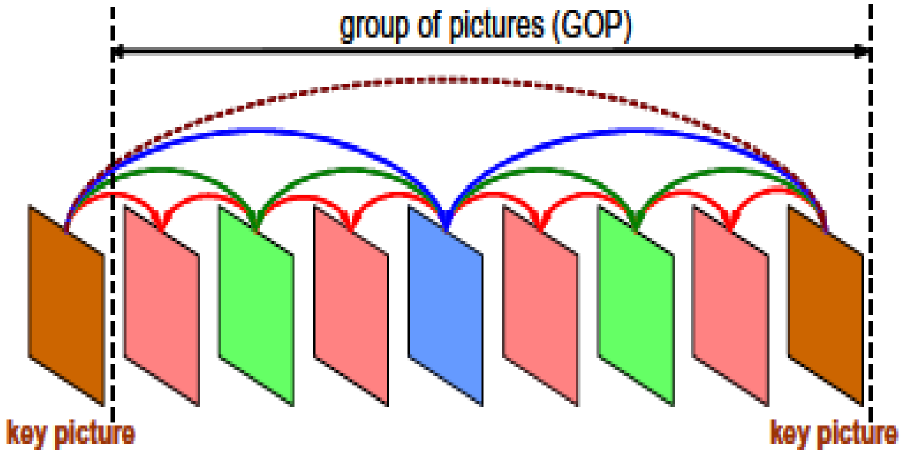

Figure 2.9: Group of Pictures (GOP) and temporal prediction in GOP [45]

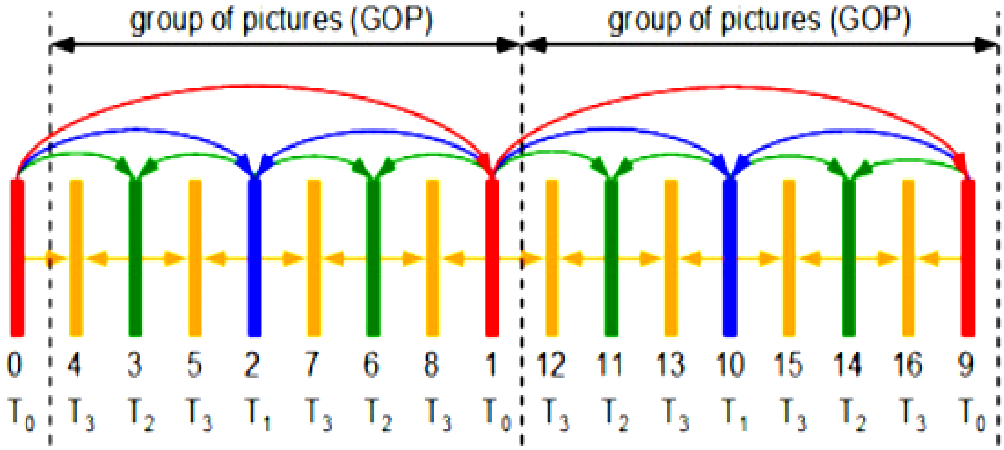

Figure 2.10: Temporal scalability [45]

B. Spatial Scalability

For supporting spatial scalable coding, SVC follows the conventional approach of multi-layer coding, which is also used in H.262/MPEG-2 Video, H.263, and MPEG-4 Visual. In each spatial layer, motion-compensated prediction and intra prediction are employed as for single-layer coding. In addition to these basic coding tools of H.264/AVC, SVC provides so-called inter-layer prediction methods which allow an exploitation of the statistical dependencies between different layers for improving the coding efficiency (reducing the bit rate) of enhancement layers. This is shown in Figure 2.11.

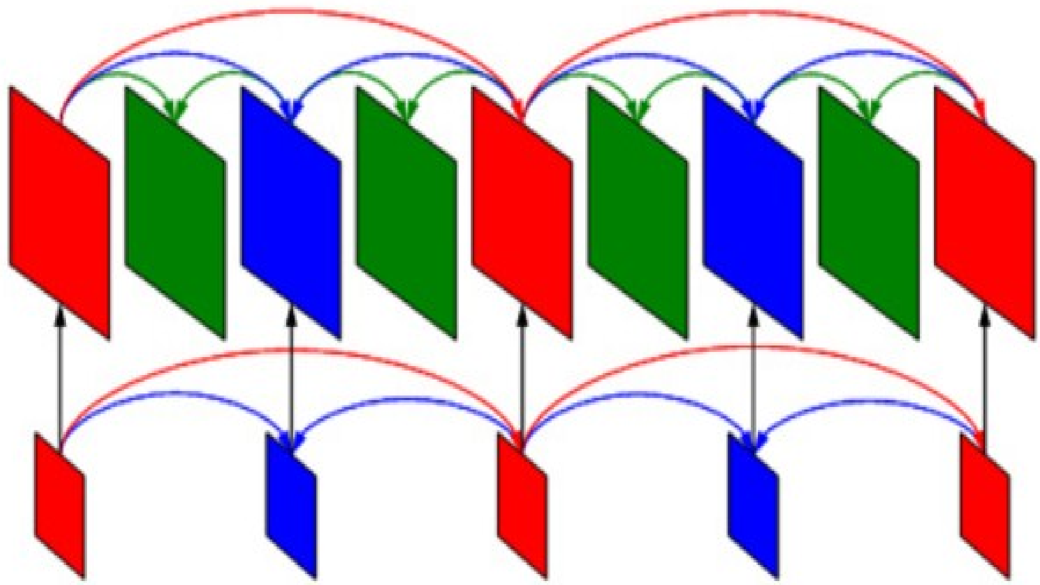

Figure 2.11: Spatial scalability [45]

C. Quality Scalability (SNR Scalability)

Quality scalability can be considered as a special case of spatial scalability with identical picture sizes for base and enhancement layer. The same inter-layer prediction mechanisms are employed, but without using the corresponding up sampling operations. Quality scalability is also called Fidelity scalability. Different quantization parameters (QP) are used in base layer and enhancement layers.

D. Combined Scalability

In the SVC extension of H.264/AVC, the basic concepts for temporal, spatial, and quality scalability can be combined. We can combine temporal, spatial and quality scalability features based on application requirements. Combined scalability can be achieved as show in the Figure 2.12. In this example, 3 temporal layers and two spatial layers are combined. Two quality layers are embedded in the first spatial layer, Layer 0.

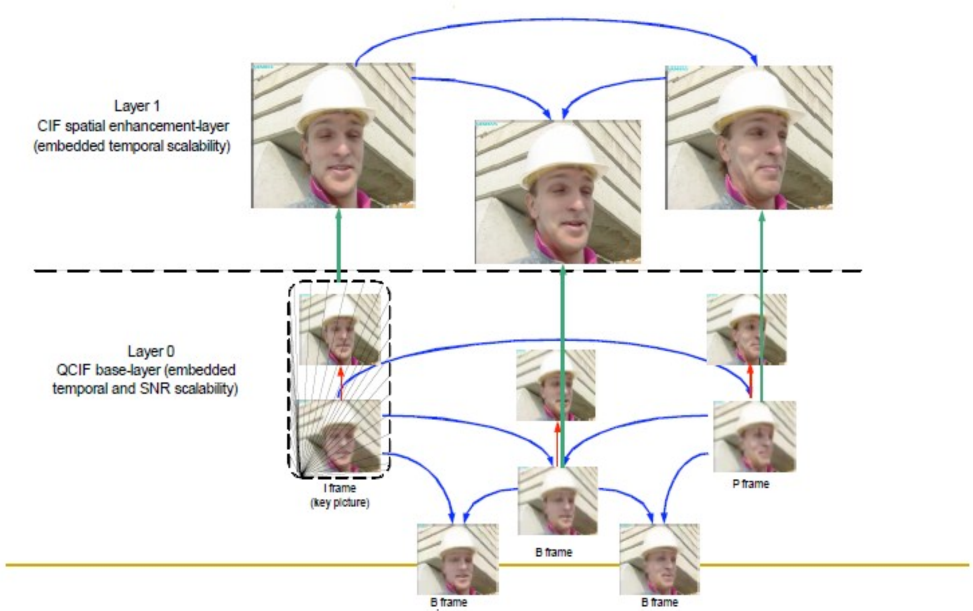

Figure 2.12: Combined scalability [45]

However, an SVC bit stream does not need to provide all types of scalability. Since the support of quality and spatial scalability usually comes along with a loss in coding efficiency relative to single-layer coding, the trade-off between coding efficiency and the provided degree of scalability can be adjusted according to the needs of an application. Following criteria's should be considered before designing a SVC system.

- Similar Coding efficiency of single layer coding
- Only little increase in decoding complexity
- Backward compatibility of the base layer
- Rate-Distortion performance comparable to non-scalable bit stream

2.4 Video Streaming over Wireless

Increased bandwidth availability in wireless networks and increased processing power of mobile handsets made video streaming over wireless with reasonable QoS a reality in the last decade. High demand for youtube like services and proliferation of new market segments like mobile TV, VoD using mobile devices and mobile video gaming raise new technical challenges for this. A lot of research is happening on new video coding standards, adaptive video streaming methods, new scheduling algorithms for delivering video with acceptable quality to most number of users. Since WiMAX was one of the first wireless networks to support high bandwidth requirements required for video streaming, a lot of research studies are conducted to support video streaming over WiMAX. System design options for video broadcasting over wireless network are studied in [7]. Major system design parameters for video broadcast over UMTS MBMS network using H.264 video coding is discussed in this paper.

2.4.1 Cross Layer Design for Video Streaming over Wireless

OSI layered approach was used successfully in wireline networks. In this approach, communication systems are organized and divided into layers. Each layer is built on top of the one below it and each layer offers services to the respective higher layer. Since each layer should fulfill a limited and well defined purpose only, this approach is having reduced complexity and it is very modular in design. The main design goal of this approach is to minimize the information exchange between different layers and dependencies between each layer should be minimal. Transmission Control Protocol (TCP)/IP protocol stack in Internet is a very good example of this approach. But, wireless networks characteristics are quite different from wireline systems. Following are the characteristics which differentiate the wireless networks:

- Noise (receiver & background)
- Higher Bit Error Rates (channel quality dynamically changes)
- Path loss
- Multipath signal propagation
- Fading effects
- Interference effects
- Mobility
- Limited resources (power and bandwidth)

These wireless channel characteristics generally affect all upper layers. Because of this, fixing problems locally inside the layers and optimizing layers independently leads to unsatisfactory results and QoS for end user. In cross layer design, dependencies between layers are exploited to fulfill QoS demands of applications. For this, layers share information with other layers to highest possible adaptivity and stability in situations of instable channel conditions. Last decade a lot of research was focused on improving the spectral efficiency of wireless medium to increase the data rate achievable. But, spectral efficiency achieved with LTE and other 4G technologies is close to Shannon limit [34]. Using the available radio resources efficiently is very important to satisfy the stringent QoS requirements of future multimedia applications like mobile TV and mobile VoD.

3GPP video services and options for cross layer design are discussed in [47]. 3GPP video services are supported by many different QoS tools, which are applied depending on network conditions and service constraints. It is possible to improve the performance of 3GPP video services tremendously by using cross layer design approach. Authors present some of the examples which uses cross layer design for performance enhancement. There are basically two different approaches for cross layer design

- Bottom up approach: Information is exchanged from bottom layers to top layers. Channel quality based adaptation use this approach.
- Top down approach: Information is exchanged from top layers to bottom layers. Video content based adaptation use this approach.

An end to end solution for efficient delivery of video broadcast over mobile WiMAX networks is presented in [53]. This paper addresses the key design issues in broadcast video streaming over WiMAX such as robust video quality, energy efficiency, and coverage. These solutions are based on cross layer optimizations. Another approach for cross layer design for wireless video streaming is discussed in [17], where video packets are prioritized in a GOP-by-GOP manner based on the dependence relation within a GOP. The priority information is conveyed in the header extension of the network abstraction layer unit (NALU) of H.264/SVC, which is parsed by a media aware network element (MANE) to realize an efficient transmission and congestion control. Cross layer design for wireless video streaming using SVC is presented by authors in [9].

2.5 Adaptive Video Streaming Algorithms Using SVC

Adaptive video streaming over wireless networks was studied from different perspectives in the past. Most of these studies were confined to mobile WiMAX networks. Mobile video transmission over wireless links using SVC is presented in [49]. The integration of SVC into mobile multimedia networks is discussed by the authors. An adaptive solution for VoD systems using SVC is discussed in [17]. A scalable video streaming system using SVC over mobile WiMAX networks is presented in [24] and [25].

A gradient based packet scheduling scheme for multiuser scalable video delivery over wireless networks is presented in [35]. In the proposed scheme, a user utility is defined as a function of some quality of service measure (such as throughput); and then maximize a weighted sum of the user's data rates where the weights are determined by the gradient of the utility function. A content-dependent packet dropping strategy that prioritizes the available packets based on their contribution to the overall quality of the video signal is used by the authors. Coded video data of the SVC are organized into packets with an integer number of bytes, called Network Abstraction Layer (NAL) units. Each NAL unit belongs to a specific spatial, temporal, and quality layer. Each video packet contains information about its own decoding deadline. The decoding deadline of a packet stems from the video streaming requirement that all the packets needed for the decoding of a frame of the video sequence must be received at the decoder buffer prior to the playback time of that frame. Any packet that is left in the transmission queue after its decoding deadline has expired is dropped since it has lost its value to the decoder. Since the base layer of the key picture is required for the decoding of all pictures of the GOP, it is given the highest priority and thus is the first packet to be added in the queue for transmission. Subsequent packets are ordered such that the next highest priority is given to the decodable packet.

Another scheduling and resource allocation algorithm, which prioritizes the transmissions of different users by considering video contents, deadline requirements, and transmission history, is proposed in [23]. Time, frequency and multi-user diversities of the OFDM system are utilized for the proposed method. Authors of [31] present a delay and capacity constrained multi-user scalable video streaming scheme that improves the average end-to-end distortion of transmitted video streams compared to traditional streaming strategies. Scalable video streaming traffic delivery in IP/UMTS networking environments is discussed in [39]. Based on the content of each packet, priorities are assigned according to the anticipated loss impact of each packet on the end-to-end video quality in this paper. Each layer has a priority range, and each packet has different priority according to its payload. The packets that contain data of an I-frame are marked with lowest drop probability, the packets which contain data of a P-frame are marked with medium drop probability and the packets which contain data of a B-frame are marked with high drop probability.

The design and analysis of channel-aware schedulers for unicast have received significant interest within the research community. But, with the proliferation of mobile TV services, video multicast is expected to increase its share in the traffic load of cellular networks. New scheduling algorithms for multicast video delivery of SVC video stream in wireless networks are studied in [51]. The users' utilities in scheduling schemes are designed as functions of the video distortion and the weights are modified to take into account the play out deadlines of video packets. Packets within a stream are prioritized based on their importance (a packet is considered more important if its loss would cause larger increase in the video distortion). If resource allocation in a Multicast group is static, resources will be wasted or call will be blocked resulting in lower quality. A dynamic bandwidth allocation scheme for mobile WiMAX networks based on SVC is discussed in [16]. This new algorithm proposes methods to solve this problem. New calls, handoff calls, and moving calls are taken into account for the radio resource management strategy to achieve optimum results. Based on the received SNR of each Subscriber Station (SS), the Base Station adjusts the burst profile using SVC codec to improve the visual quality and system throughput. The traffic is processed by a weighted round robin scheduling algorithm once the call is accepted.

Similar to adaptive scheduling, adaptive modulation techniques are important for delivering different layers in SVC with different importance. A simple, but very useful AMC scheme is presented in [11], where unused bits in header of RTP,UDP and IP packets are used to carry SVC layer information (to identify the SVC layer carried in the packet) to the MAC layer. This information is used by the MAC layer to distinguish the received SVC layer in the packet and take decision regarding modulation to be used.

Chapter 3

SVC Based Adaptation Scheme for Video Transport over LTE Networks

3.1 Introduction

Video transmission with the efficient use of expensive radio resources is very important if one wishes to support more mobile users with acceptable QoE. Video encoders and cross layer design in LTE access networks play a crucial role in delivering video with the least possible bandwidth and with acceptable quality. Advantages of SVC over H.264 video coding are analyzed and adaptive video streaming using the scalability features of SVC encoded video for LTE networks are studied using real time use cases in this chapter.

3.2 Analysis of SVC Based Adaptation Scheme for Unicast and Multicast

This chapter analyzes the advantages of using SVC over H.264 video coding standard for video streaming over LTE networks. A simple, but useful scheme for SVC video streaming is also proposed here. Proposed scheme covers different scenarios in multicast and unicast video transmission in LTE networks. This scheme reduces the packet loss ratio with considerable reduction in bit rate, as compared to H.264 based video streaming solutions. The adaptation scheme is shown in Figure 3.1 and is briefly described below.

- When a mobile video streaming session is requested, the decision about multicast or unicast transmission will be made based on the number of users requesting the same video. If multiple users are requesting the same video (e.g. Mobile TV), multicast is selected.
- Spatially scalable video is sent from the video server in multicast case to support heterogeneous receivers in the multicast group. CIF or QCIF resolution videos are sent in the unicast case, depending on the UE display type.
- Users near the base station are allocated to Multicast Group1 and users far from the base station are allocated to Multicast Group2. Temporal layers are dropped in the base station for Multicast Group2.

- In the unicast case, quality layers are dropped in the base station based on the channel quality, congestion in the base station or UE battery/processing power.

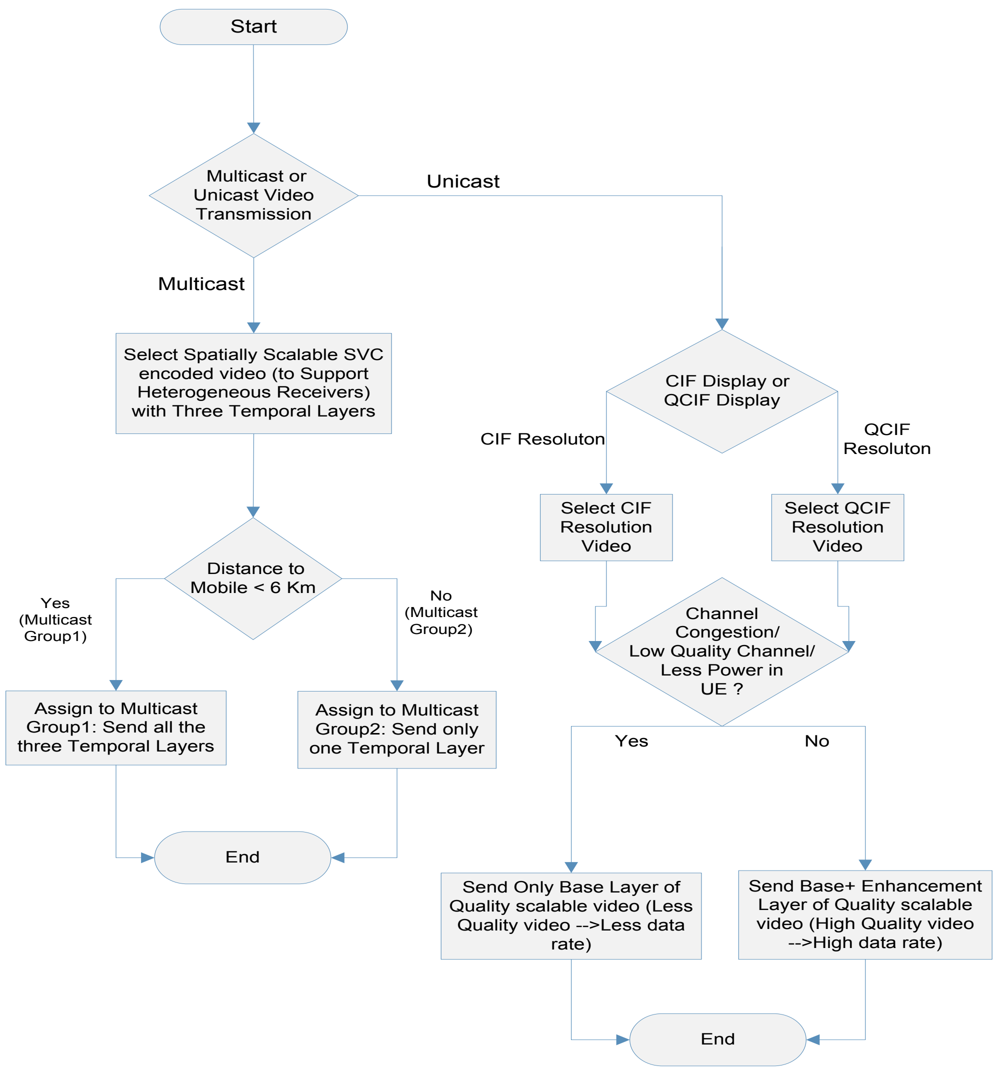

Figure 3.1: SVC based adaptation scheme

Three different scenarios to verify this scheme are briefly described below.

3.2.1 Heterogeneous Receivers with Multicast

This use case analyzes the advantages of using SVC over H.264 for supporting receivers with different display resolution capabilities in multicast scenarios. With H.264 single layer video, simulcast of two different resolution videos are required to support two different resolution mobile devices. But, SVC allows multicast of multilayer video with Quarter Common

Intermediate Format (QCIF)/Quarter Video Graphic Array (QVGA) base layer and Common Intermediate Format (CIF)/Video Graphic Array (VGA) enhancement layer. The throughput requirement of multicast video with SVC is supposed to be considerably less when compared to simulcast of two H.264 videos.

3.2.2 Graceful Degradation in Multicast

Individual feedbacks for UEs are not available in multicast scenarios. Therefore, a static algorithm for bit stream adaptation is required. Generally, users near the base station are assumed to be experiencing good channel conditions and, users in cell edge are assumed to be experiencing bad channel conditions. Using H.264 encoding, the same video needs to be multicast to all the users in the cell. This will result in abrupt quality degradation for users in the cell edge. However, using the quality and temporal scalability features of SVC, graceful degradation of video quality can be provided. Since distribution of mobile display resolution does not follow a specific pattern inside the cell, spatially scalable video need to be sent from the video server. Separate multicast groups for mobiles with the same display resolution is another possible option for handling this scenario.

In our simulations, input video is encoded with spatial and temporal scalabilities with QCIF base layer and CIF enhancement layer. All the three temporal layers are sent to users near the base station (defined as < 6Km and named Multicast Group1). Users far from the base station (defined as > 6Km and named Multicast Group2) are served with only temporal layer 0, which contains only the Intra coded frames (I Frames) and Predictive coded frames (P frames). Layers 1 and 2 of temporal scalability contain Bidirectional predicted frames (B Frames). The dropping of B frames does not cause abrupt video quality fluctuations and this approach reduces the possibility of dropping the I and P frames. Moreover, throughput saving in the eNodeB is expected with this approach.

3.2.3 UE and Network Aware Streaming in Unicast

For each UE, individual feedbacks are available, and this makes the adaptation easier compared to multicast scenarios. SVC bit streams can be adapted based on UE battery power, UE processing power, Channel Quality Indicator (CQI), network congestion and display resolution of UE. Different types of scalabilities can be applied independently or combined, based on the requirements of the adaptation. In our experiments, quality scalability is used for sending low quality videos to UEs with less channel quality (at 10 Km) and high quality videos to UEs with high channel quality (at 5 Km).

The simulation model of the LTE network is created with one video server streaming different videos, and heterogeneous receivers distributed randomly inside the cell coverage area. Network topology used for testing is given below. System configuration used in the simulation is given below in Table 3.1.

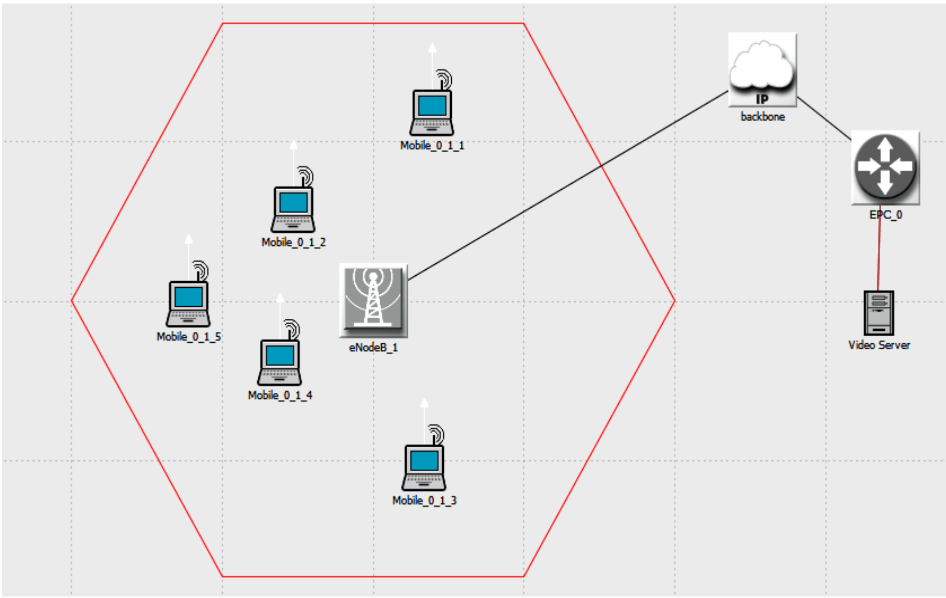

Figure 3.2: OPNET LTE simulation model

Table 3.1: LTE system parameters

Parameters	Value
LTE duplex mode	FDD
Traffic direction	Downlink
Carrier frequency	2.5 GHz
LTE bandwidth	20 MHz
MAC scheduler	Link and channel dependent
User location	Randomly distributed (and based on the use case)
Path loss model	Free space
Multipath channel model	ITU Pedestrian A
# of Tx and Rx Antennas	1
Radio Link Control (RLC) mode	Unacknowledged Mode (UM)
Maximum number of HARQ retransmissions	3
eNodeB operating power	46 dBm

H.264 and SVC video traces available in [50] are used for the simulation. Simulations are conducted using two different videos to confirm the validity of the results. Video traces of Tokyo Olympics and NBC News videos were used for comparison between H.264 and SVC. The results are given for Tokyo Olympics video. GOP size is selected as 16 for both video sequences. Even though QCIF and CIF video traces are used for the simulations, the same results hold for QVGA/VGA and other resolutions too. The video streaming performance between the video server and the mobile client is compared for H.264 and SVC videos using the performance metrics given below.

- Throughput: Average throughput in eNodeB (in bits/sec)
- Packet Loss Ratio: Number of packets lost compared to sent packets per second
- Video frames end-to-end delay: Delay for video frames to reach UE from video server. This includes processing delay, queuing delay and transmission delay.
- Jitter: Variation in packet arrival time

PSNR is one of the most popular objective quality measures used for video quality. But, PSNR cannot show properly the effect of predicting errors (lost I- and P-frames) that may be highly visible but that create little distortion in terms of mean square error. Lost I frames will have more of an impact on the video quality degradation, as compared to lost P or B frames. As explained in [13] and [18], PSNR cannot be used for quality prediction in cases of lost video frames over networks. However, it can be used as a quality indicator of the encoded input video for simulation. A heuristic mapping of PSNR to Mean Opinion Score (MOS) is presented in [37] and given below in Table 3.2.

Table 3.2: PSNR to MOS mapping [37]

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

Videos with Excellent and Good MOS scores are used as input video traces. In these adaptations, we are trying to reduce the number of lost frames with minimum bandwidth possible for an acceptable input PSNR video.

3.3 Simulation Results and Analysis

Simulation results are given and analyzed below. Simulations of each scenario are run for 5 times and average results are used for generating the graphs. Simulation results for each of the 5 runs were similar with very minor variations because of the multipath.

3.3.1 Heterogeneous Receivers with Multicast

Average throughput, delay and jitter for H.264 simulcast and SVC multicast are compared below. Average throughput in the MAC (Medium Access Control) layer of eNodeB is compared in Figure 3.3 and average jitter is compared in Figure 3.4. Since delay and jitter in the core network is not modeled realistically, these values will change in real networks. The values of delay and jitter given below are given only for the comparison of different encoding methods.

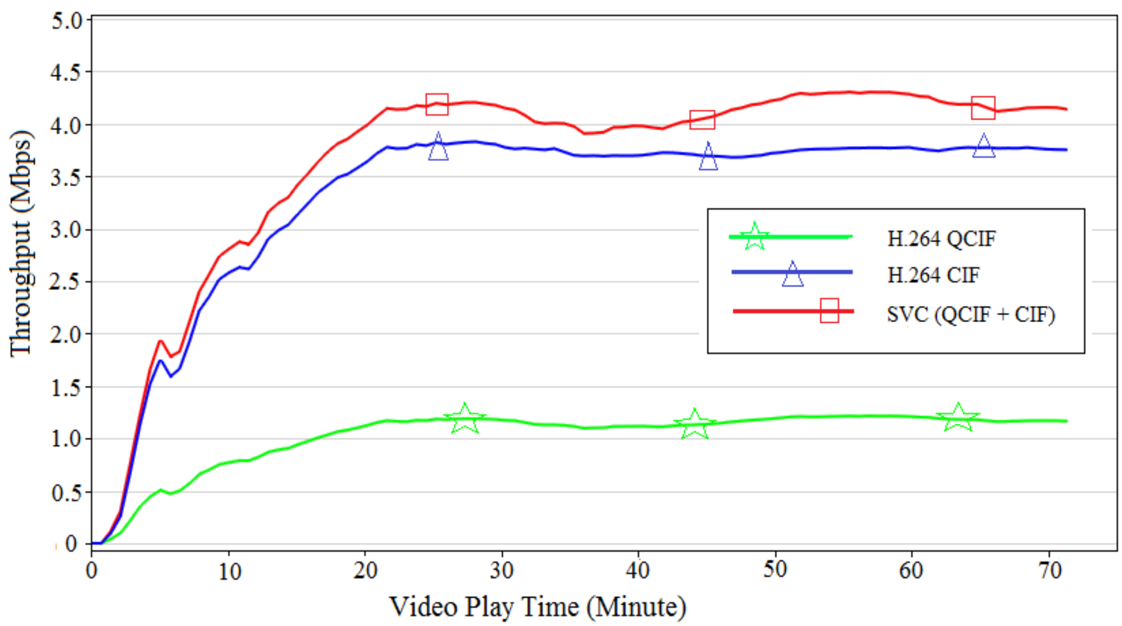

Figure 3.3: SVC and H.264 average throughput comparison

The results are summarized in Table 3.3.

Table 3.3: Comparison of H.264 and SVC videos

Parameters	H.264 CIF	H.264 QCIF	SVC CIF + QCIF
PSNR (dB) [50]	50.37	32.11	50.81
Throughput	3.8 Mbps	1.2 Mbps	4.1 Mbps
Delay	4.3 ms	6.5 ms	7 ms
Jitter	27 us	22 us	34 us

From the simulation results given above, we can see that SVC multicast requires considerably less data rate (4.1 Mbps) compared to the simulcasting of two H.264 videos (5 Mbps). There is almost an 18% reduction in bit rate achieved with SVC multiple layer video in this case. There is a small increase in the delay and jitter due to SVC encoding of the video. But, effect of this small increase in delay and jitter can be easily controlled using decoder buffer and jitter buffer in the receiver.

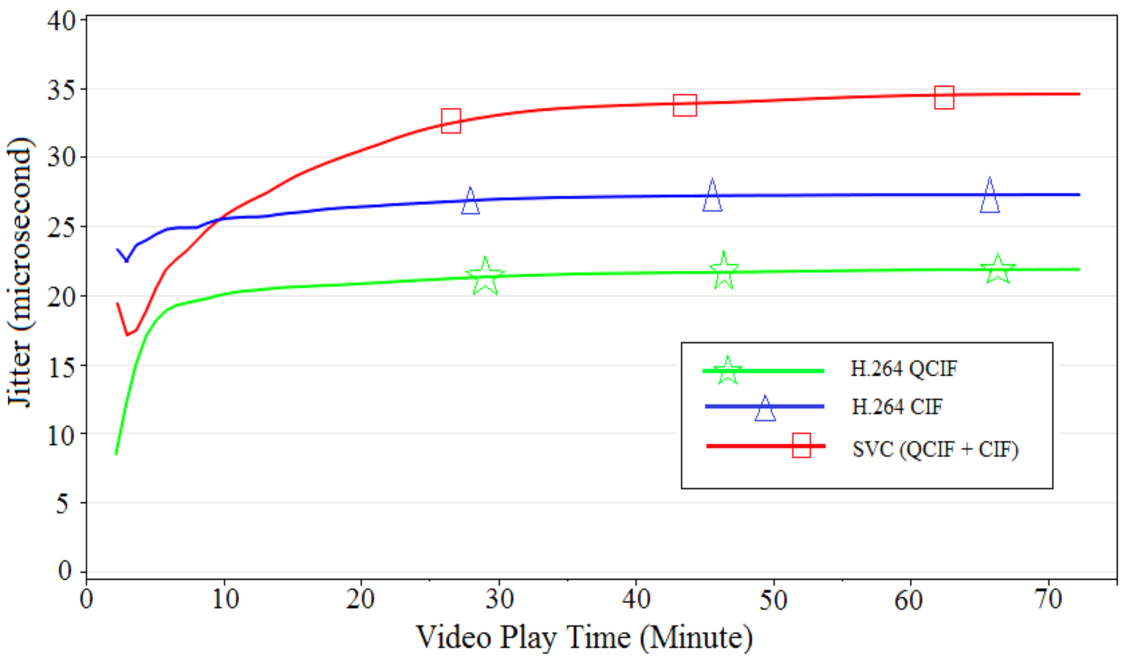

Figure 3.4: SVC and H.264 jitter comparison

3.3.2 Graceful Degradation in Multicast

The video frames received per second for H.264 encoded videos by mobiles located 5 Km and 10 Km from the base station are shown in Figure 3.5. In the lossless situation, 30 frames are supposed to be received in the UE video processing application; because of the 30 fps (frames per second) encoding rate in the video server. Based on this understanding, packets lost per second in the video application can be easily calculated from Figure 3.5. For users at 10 Km, on average 3 video frames per second are lost. That means, the packet loss for the 10Km scenario is very abrupt and there is a high chance of dropping the Intra Coded Frames (I- Frames), seeing as all kind of frames are sent with equal priority.

Figure 3.6 shows the packet loss pattern with graceful degradation applied by SVC encoding. According to the proposed method, all the three temporal layers are sent to user in 5Km radius (Multicast Group 1) and only base temporal layer is sent to two users in 10 Km radius (Multicast Group 2). We can see that all the UEs have equal chances of packet loss and the packet loss ratio is reduced for UEs at 10 Km. In our simulations, only temporal scalability is used for adaptation purpose. However, quality scalability or combination of quality and temporal scalability features of SVC also can be used for graceful degradation in multicast scenario.

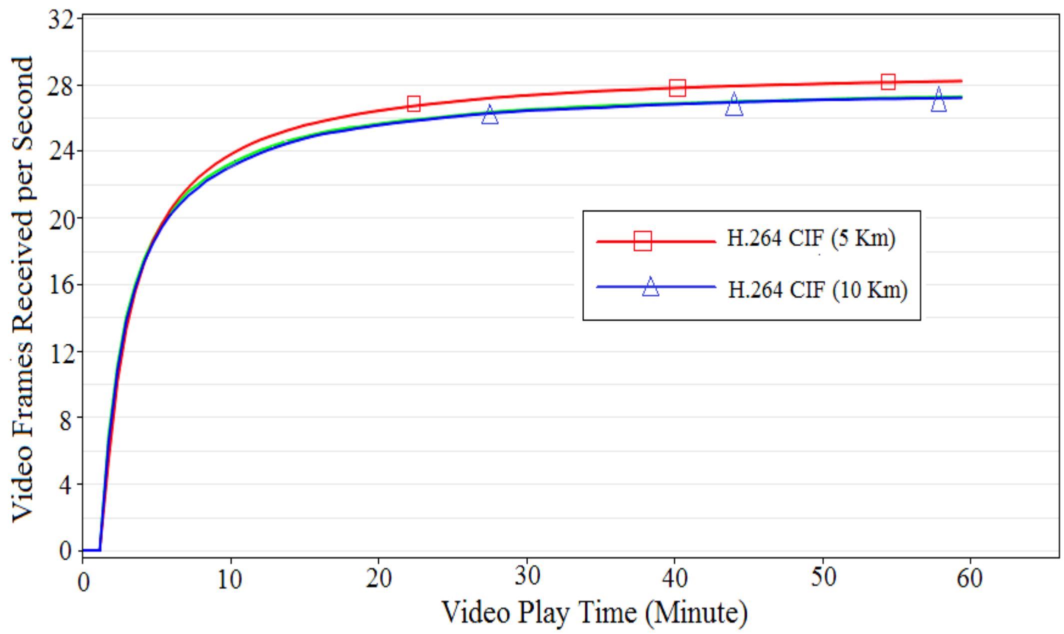

Figure 3.5: Packet Loss Pattern of H.264 Videos for UEs Located at 5 Km and 10Km from Base Station

The results from Figures 3.5, 3.6 and 3.7 are summarized in Table 3.4. The results indicate graceful degradation in quality for two SVC users at 10 Km with reduction in average throughput and packet loss ratio, but with acceptable PSNR value. Even though we analyzed only temporal scalability for graceful degradation, quality scalability and temporal scalability can be applied simultaneously for this kind of graceful degradation.

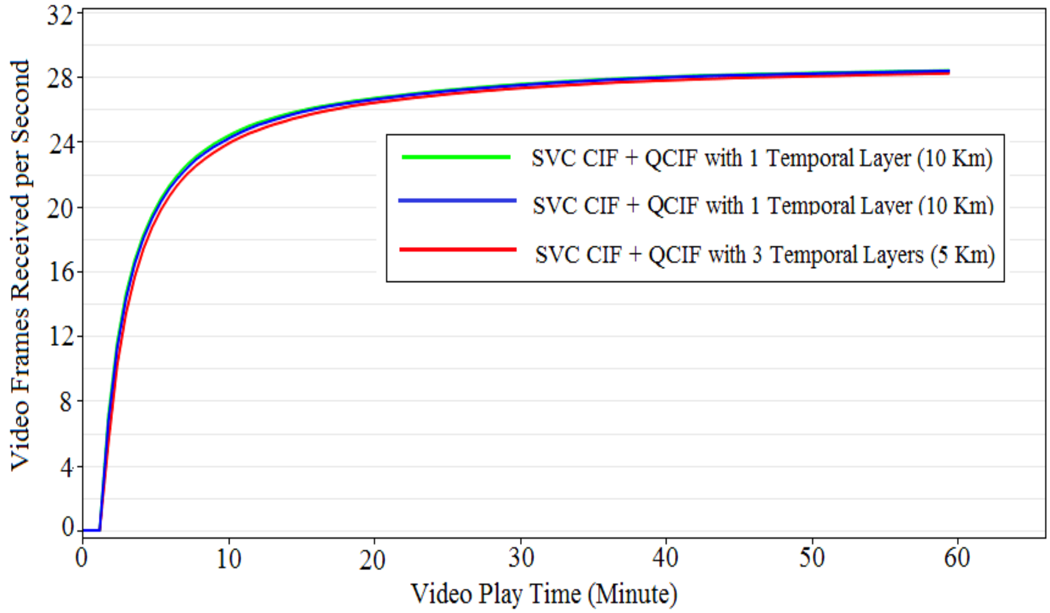

Figure 3.6: Packet loss pattern of SVC encoded videos for UEs located at 5 Km and 10 Km from base station (With graceful degradation)

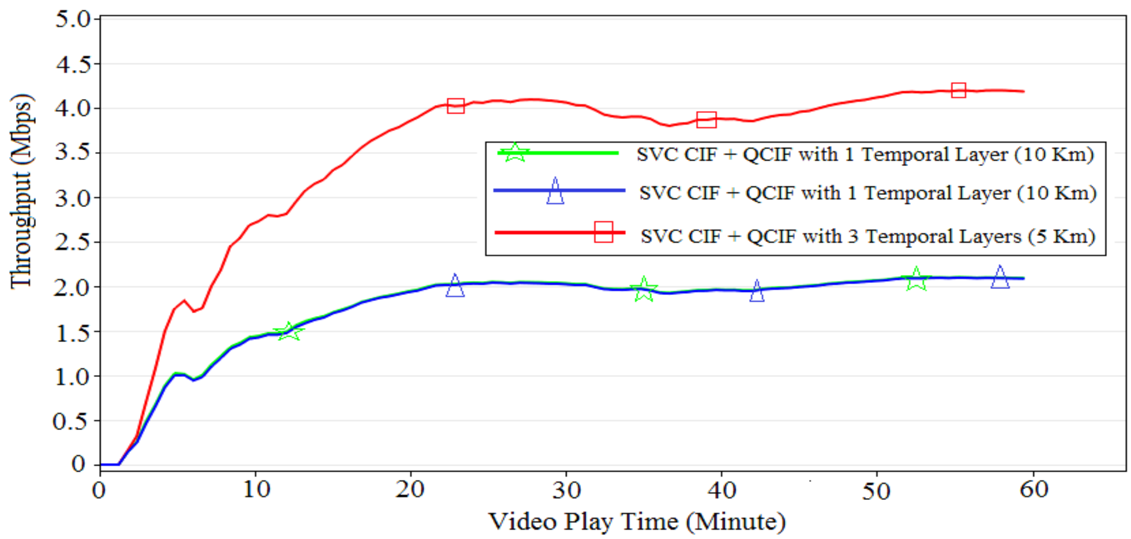

Figure 3.7: Average throughput comparison with SVC temporal scalability

Table 3.4: Comparison of H.264 and SVC with Graceful Degradation

Parameters	H.264 CIF (10Km)	SVC CIF + QCIF - 3 TLs (5 Km)	SVC CIF + QCIF - only TL0 (10 Km)
PSNR (dB)* [50]	50.37	50.81	35.82
Packet loss ratio	3/30	2/30	2/30
Throughput	3.8 Mbps	4.1 Mbps	2 Mbps

*PSNR values are given for the input video sequences and given for comparing quality degradation with temporal scalability of SVC.

3.3.3 UE and Network Aware Streaming in Unicast

Quality scalability using different Quantization Parameters is used for simulation purpose. Quantization parameter 10 is used for high quality video and 16 is used for low quality video. Video Frames received per second is shown in Figure 3.8 and throughput comparison is shown in Figure 3.9.

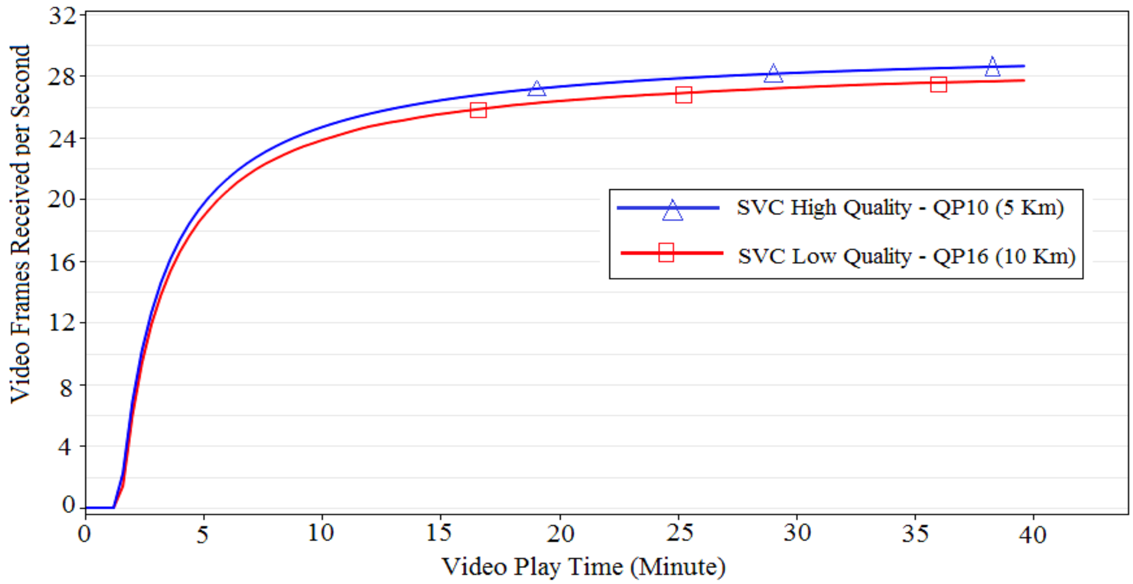

Figure 3.8: Packet loss ratio comparison for SVC quality scalability

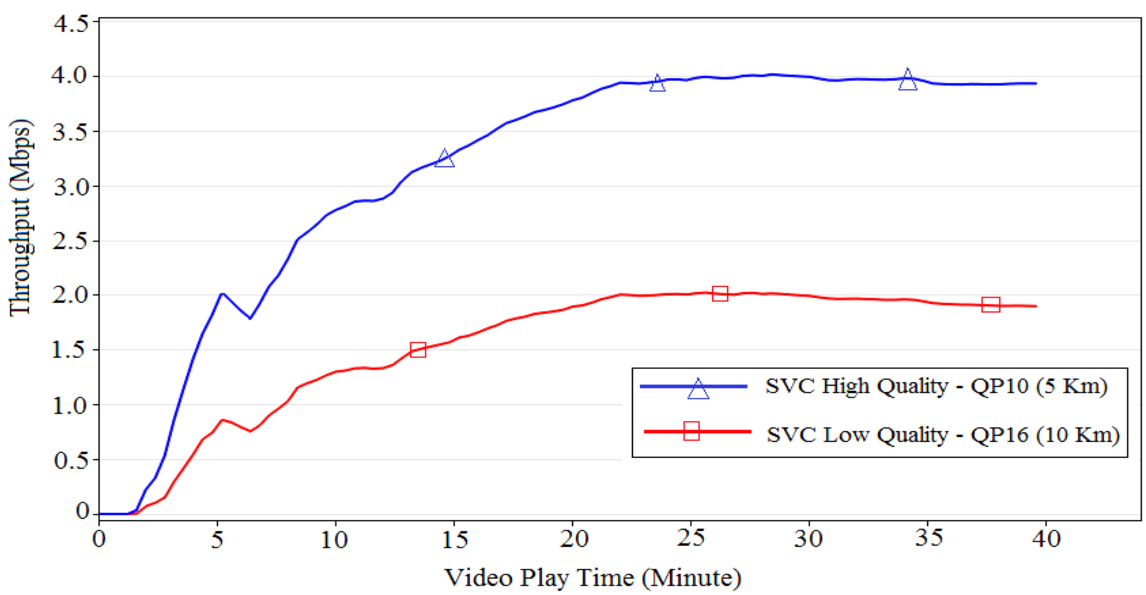

Figure 3.9: Throughput comparison for SVC quality scalability

Results are summarized in Table 3.5. Simulation results indicate considerable reduction in bit rate with very less and acceptable quality degradation using quality scalability for UE at 10Km.

Table 3.5: Adaptation for unicast streaming using SVC quality scalability

Parameters	SVC (BL + EL) QP10 (5 Km)	SVC BL (QP16) (10 Km)
PSNR (dB) [50]	51.62	46.97
Packet loss ratio	1.5/30	2/30
Throughput	4 Mbps	2Mbps

3.4 Conclusion

The advantages of using SVC in LTE Multicast and Unicast scenarios in LTE networks are obvious from the analysis above. Results show that considerable packet loss reduction and throughput savings (18 to 30%) with acceptable video quality are achieved with proposed scheme based on SVC compared to H.264. But further work needs to be done for the adaptive scheduling of SVC bit streams based on Channel quality feedback, user power level and based on the video content itself. These adaptive scheduling algorithms should be able to map SVC video layers to physical and link layer resources of LTE in optimized and efficient manner.

Chapter 4

Cross Layer Design for Unicast Video Streaming Using SVC over LTE networks

4.1 Introduction

Channel dependent adaptations are usually done in the MAC layer because of the unavailability of channel state information in the video server. To overcome this limitation, we propose a cross layer signaling between MAC layer in the eNodeB and RTP layer in the video server. Average CQI values are signaled from MAC layer to RTP layer, and this information is used for adaptation by the RTP layer. Scalability features of scalable video are used for efficient adaptation in the video server. To exploit the frequency diversity of the channel, channel dependent MAC scheduling is used in the MAC layer. In addition, we propose a new scheme for adaptive GBR selection using CQI of the user. Our adaptation scheme is applicable for unicast video on demand delivery systems with LTE wireless access. Delivering acceptable Quality of Service (QoS) to the maximum number of users in a cell coverage area with minimum usage of radio and other network resources is the goal of this adaptation scheme.

4.2 Proposed Adaptation and Scheduling Scheme for Unicast SVC Video Transmission

Channel quality based adaptive bit rate selection and scheduling scheme for video transmission over LTE network is presented here. Adaptive bit rate is achieved using quality and temporal scalability features of SVC encoding. CQI reports from UEs are used for the adaptation purpose, and this scheme is aimed at improving video quality of low and medium quality users and reducing the bandwidth consumption of expensive wireless spectrum. Packet losses in an end to end video traffic system with LTE wireless access can be of three types: packet losses in backhaul router due to congestion, packet losses in eNodeB due to congestion in the eNodeB buffer, and packet losses in the wireless medium. Packets sent from eNodeB to wireless medium, which will be dropped in the wireless medium due to bad channel quality or will be dropped at the receiver due to less processing power or battery power, is a waste of radio resources.

UEs in LTE network update eNodeB about the channel quality using CQI reports. For adapting to fast channel quality variations, periodic CQI reporting scheme is used with a reporting interval of 2ms. The MAC layer in the eNodeB provides CQI information for each user to the RTP layer in the video server. Since RTP is not aware of resource management in the frequency spectrum, sub-band CQIs are not useful for adaptations in RTP layer. So, wideband CQI values are used for the adaptation purpose in video server. Since adaptations for each and every CQI is not practical and not possible to implement in real time systems, adaptations are performed over a group of CQIs. Our proposed adaptation scheme is shown in Figure 4.1

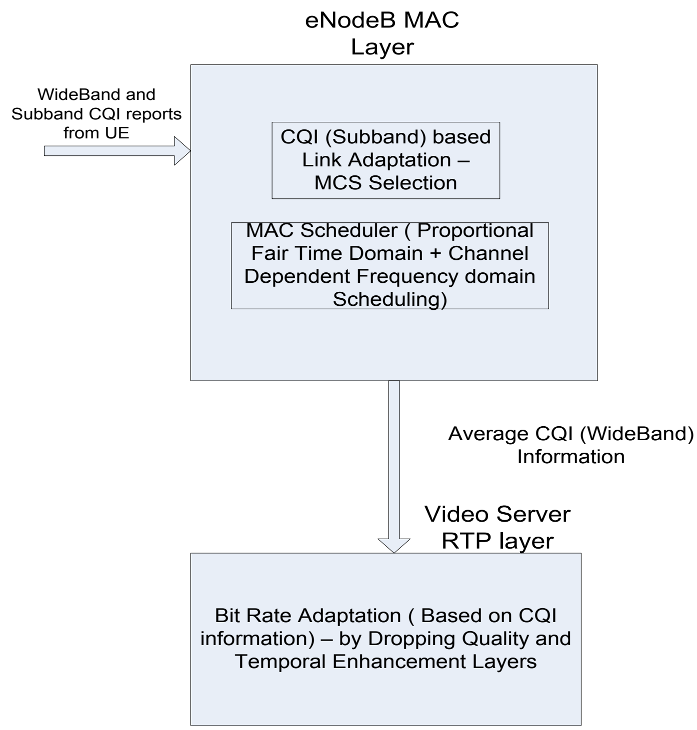

Figure 4.1: Channel quality based bit rate adaptation in the video server

Based on the CQI, MCS value for the user is changed dynamically. A two-level scheduler is used in the eNodeB MAC to ensure fairness among users and to exploit channel diversity of users. A proportional fair time domain scheduler is used for video bearers, and channel state dependent frequency domain scheduling is used to exploit multi-user diversity in the fading wireless medium. Sub-band CQI reports are used in the scheduler to allocate preferred sub-band to each user. Proposed scheduling scheme in the MAC layer is shown in Figure 4.2.

SVC adaptation scheme used for different channel qualities are as shown in Table 4.1. Quality and temporal scalability features of SVC bit stream are used for adaptation. Since unicast

VoD systems require transmission of single resolution video based on UE type, spatial scalability cannot be used here.

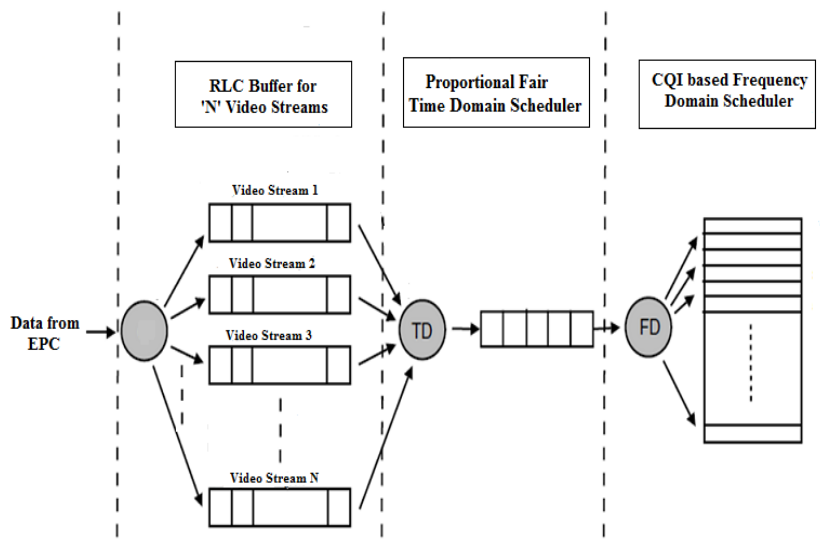

Figure 4.2: MAC scheduling scheme in the eNodeB

Table 4.1: SVC adaptation scheme in the video server

CQI Range	**SVC Adaptation**
10 – 15 (High quality channel)	Three quality layers and three temporal layers
7 – 9 (Medium quality channel)	Two quality layers and two temporal layers
1 – 6 (Low quality channel)	One quality layer and one temporal layer

Bit rate possible for channel quality of the user is measured statically and used in this dynamic adaptation. QP for base quality layer is selected based on the bit rate possible for low quality user. Similarly, QPs for two enhancement quality layers are selected based on the bit rate possible for medium and high quality users.

Our scheme is also aimed at avoiding the over provisioning of radio resources in eNodeB. Since SVC video encoding produces highly variable bit rate traffic, assigning a high GBR value is the only solution for avoiding buffer overflow in eNodeB during congestion. However, instead of assigning same GBR values to all users in the cell coverage area, we suggest the use of CQI value of UEs for assigning GBR values. This helps to assign only the required resources to each user, and valuable wireless spectrum resources can be saved. Statically measured achievable bit rate for each CQI range (in specific channel conditions) is used as the GBR value for users.

4.3 Simulation Model

Since adaptations happening in the video server and scheduling in eNodeB are studied here, single cell network is used for the simulation purpose. Infinite bandwidth model is used in the core network links and routers. Network topology used for simulations is shown in Figure 4.3 and LTE system configuration used in the simulation is given below in Table 4.3. Simulations are done using Star Wars 4 movie video trace from [50]. Since LTE needs to support heterogeneous receivers including high resolutions with high quality video, high quality SVC video stream is used for the analysis purpose. SVC encoder setting used is shown in Table 4.2. Rate distortion curve for Star Wars SVC bit stream is shown in Figure 4.4. Since rate distortion curve is not linear, reduction in bit rate will not affect video quality linearly. Rate distortion curve is non-linear for all other videos available in [50] and it is a general property for a SVC bit stream.

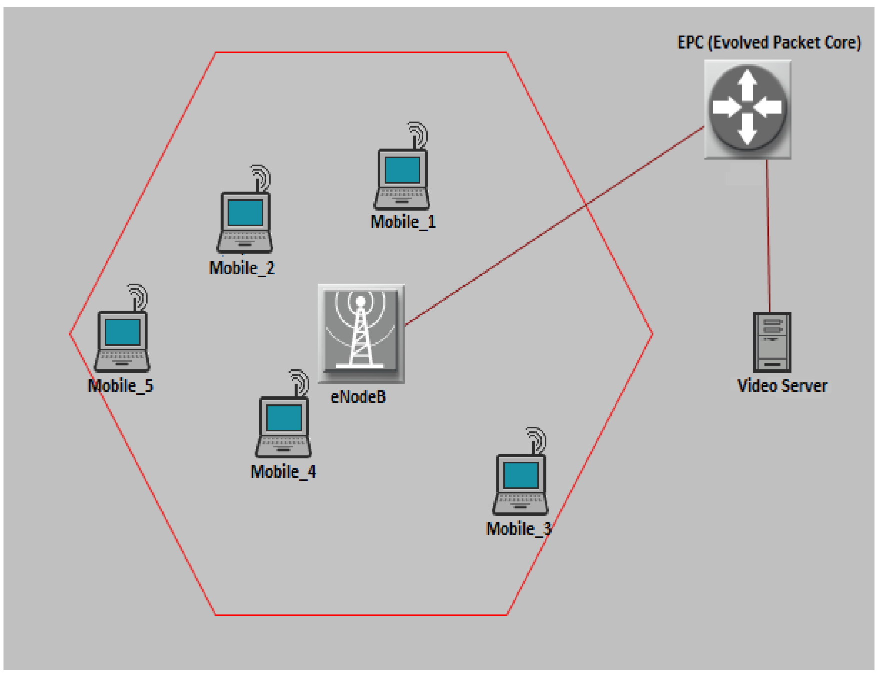

Figure 4.3: OPNET simulation model

Table 4.2: SVC encoder settings

Parameters	Value
Number of Frames	500
Resolution	CIF (352 X 288)
Frame Rate	30 fps
GOP	16
Number of B Frames between I and P Frames	3
Encoding Bit Rate	Variable Bit Rate (VBR)
GOP Structure	IBBBPBBBPBBBPBBB
SVC scalability layers	3 Quality and 3 Temporal Layers

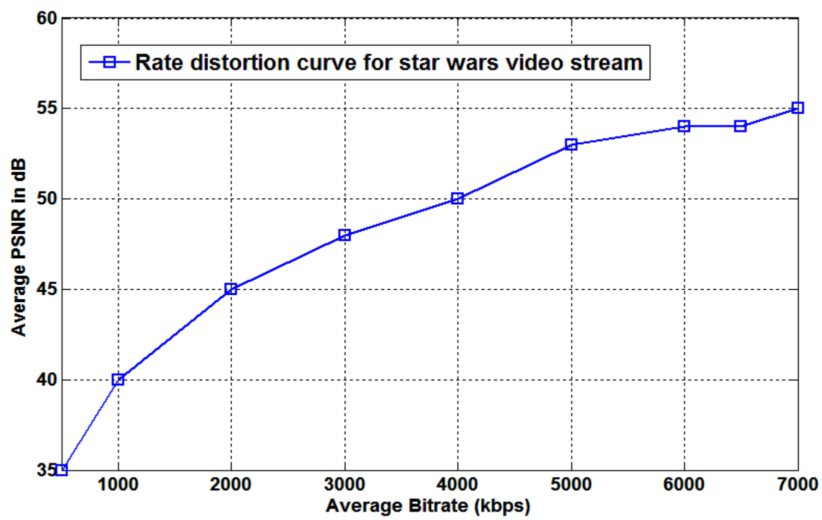

Figure 4.4: Rate distortion curve for star wars movie video bit stream

As explained in Section 3.2, PSNR cannot be used to evaluate the video quality of SVC video. To evaluate the quality of an output video sequence, we adopt an evaluation model of video quality (Q) from [29], which is defined by the number of decodable frames over the total number of frames originally sent from the video source.

$$Q = \frac{N_{dec}}{N_{total-I} + N_{total-P} + N_{total-B}}, \quad (4.1)$$

Where $0 < Q < 1$, N_{dec} is the total number of decodable frames including all three types of frames (i.e., $N_{dec} = N_{dec\text{-}I} + N_{dec\text{-}P} + N_{dec\text{-}B}$). It is assumed that dependencies between different types of frames (i.e., I, P, and B frames) are already considered in the derivation of the numbers of different decodable frames. $N_{total-I}$, $N_{total-P}$ and $N_{total-B}$ are the total number of I-, P-, B-frames in the video source, respectively. I (Intra-coded) frames are encoded independently and decoded by themselves. P (Predictive-coded) frames are encoded using predictions from the preceding I or P frame in the video sequence. B (Bi-directionally predictive-coded) frames are encoded using predictions from the preceding and succeeding I or P frames. The loss of an I frame is essentially equivalent to the loss of all the frames in the GOP. B frames are not used for decoding any other frames. The above mentioned dependencies are used in the calculation of N_{dec-I}, N_{dec-P}, and N_{dec_B}. Q is an objective measure to evaluate the video quality. The larger Q means the better video quality is perceived by the recipient. Since our adaptation method uses different bit streams for throughput and quality adaptations, this objective quality measure needs to be multiplied with input PSNR value to get a reliable objective quality measure for output bit stream.

In other words, Output video quality measure = Q * $PSNR_{input\_video}$

Table 4.3: LTE system parameters

Parameters	Value
LTE mode	FDD
Traffic direction	Downlink
eNodeB operating power	46 dBm
Carrier frequency	2.5 GHz
LTE bandwidth	10 MHz (50 PRBs)
User location	Users with different channel qualities (high, medium and low) equally distributed in the cell coverage area
MAC scheduler	Proposed scheduler (proportional fair in time domain and channel dependent in frequency domain)
Path loss model	Suburban macrocell
Multipath channel model	ITU Pedestrian A
# of Tx and Rx Antennas	1
CQI reporting scheme	Wideband (used for RTP adaptation) + Subbands (used for MAC scheduling) – periodic reporting scheme (2 ms interval)
RLC mode	UM
Maximum number of HARQ retransmissions	3

The following scenario is simulated to prove the benefits of CQI based adaptation in the video server. Suburban macrocell scenario with 3 km cell radius is used for simulations. Three users from different channel quality regions with CQI Low (3-4), CQI Medium (7-8) and CQI High (12-13) are used in the analysis. Distributions of these three users are as given below in Table 4.4.

Table 4.4: Mobile user distribution

User	Distance from eNodeB (Km)	Best MCS wideband index	CQI
Mobile_1	0.2 – 0.5	23 – 24	12 – 13
Mobile_2	1 – 1.5	13 – 14	7 – 8
Mobile_3	2.5 – 3	5 – 7	3 – 4

4.4 Simulation Results and Analysis

Simulation results for the proposed method are given and analyzed in this section. Simulations are run for 3 times and average results are used for generating the graphs. As expected, CQI values of users fluctuated over simulation time. This is mainly due to the multipath fading enabled LTE network model. This confirms our hypothesis as to why CQI adaptation is not done for each CQI feedback in the video server and why wideband CQI is used for the adaptation purpose. Because of the high CQI fluctuation, feedback to video server and adaptation in each 2 ms CQI reporting interval is not practical and impossible to implement. Because of this, we use long time average CQI values for the adaptation purpose in the video server.

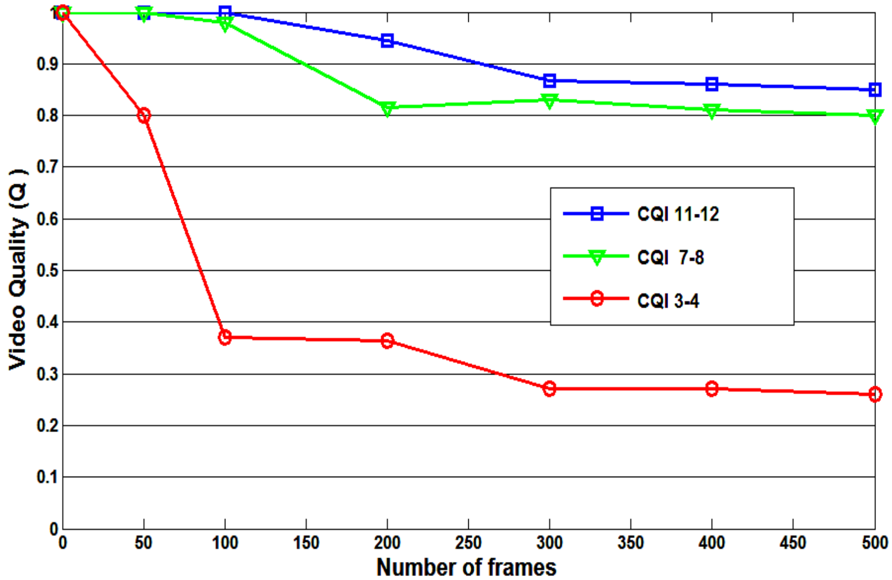

Figure 4.5: Video quality for different CQI values without adaptation

Objective video quality measured using Eqn. (4.1) for three CQI values without adaptation is given in Figure 4.5. Very low video quality for cell edge users (low CQI value) is highly noticeable. CQI based adaptation is used for medium and low quality channel users and the resulting video qualities are compared with the results obtained without adaptations in Figure 4.6 and 4.7. Objective video quality gains are very much visible from this comparison. Packet loss ratio for users without and with adaptations is plotted in Figure 4.8 and 4.9, respectively.

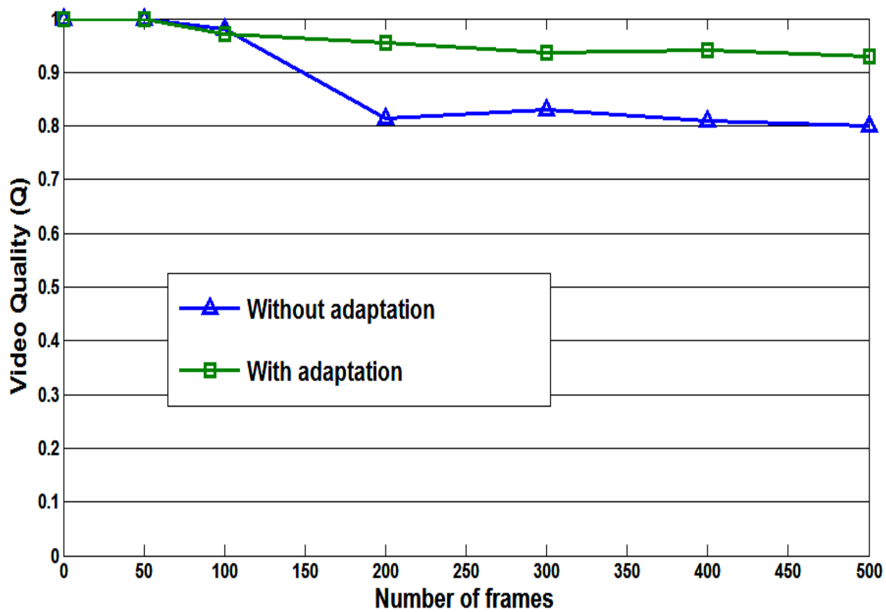

Figure 4.6: Video quality of medium CQI user (CQI 7-8) with and without adaptation

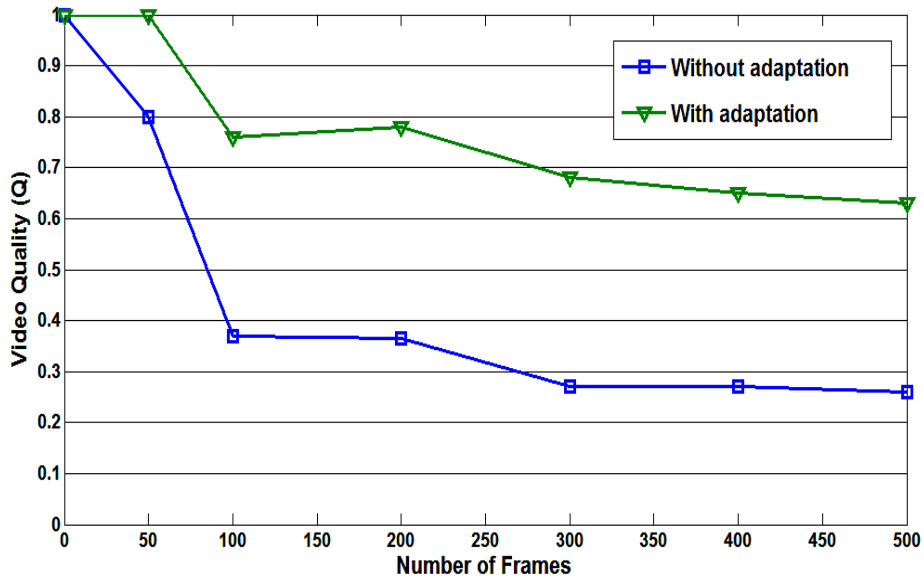

Figure 4.7: Video quality of low CQI user (CQI 3-4) with and without adaptation

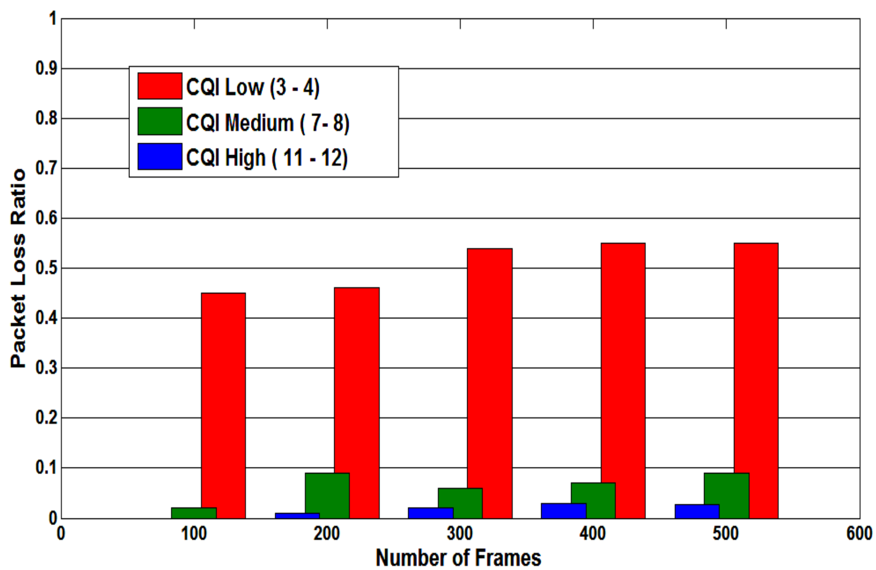

Figure 4.8: Packet loss ratio for different CQIs without adaptation

As we can see from the results, packet loss ratio is decreased and video quality is increased with the removal of quality and temporal layers based on the CQI value. Improvement of video quality for the cell edge users is very visible in Figure 4.7. Of course, this comes at the cost of reduced quality video from the video server. But, as explained and shown in Figure 4.4 in Section 4.3, video quality is not linearly reduced with bit rate reduction. This is especially true for high quality input videos. PSNR of high quality, medium quality, and low quality videos sent to three different users are shown below in Figure 4.10. This low input video quality comes at low throughput, and this reduced throughput helps to serve more users and services in the cell.

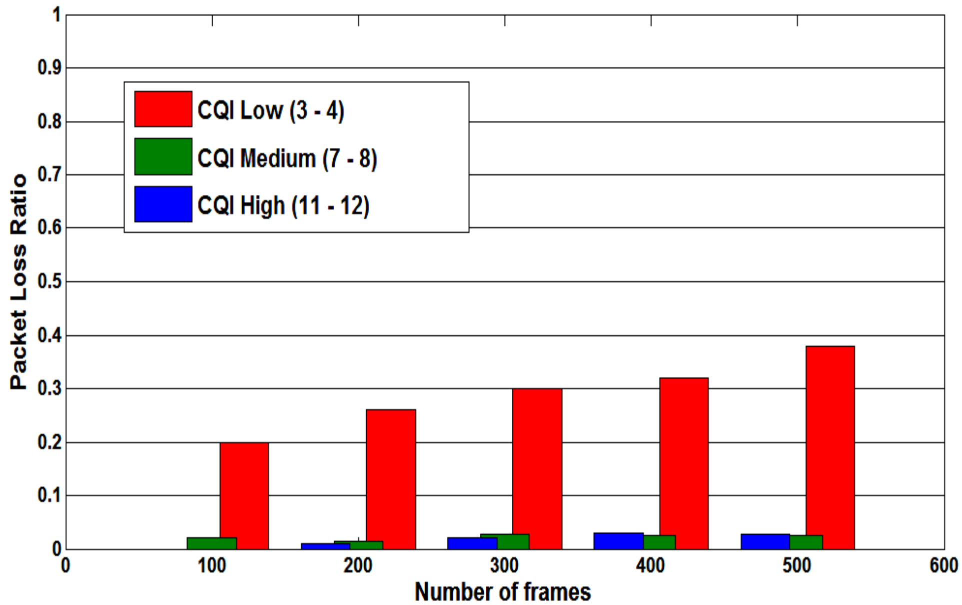

Figure 4.9: Packet loss ratio for different CQIs with adaptation

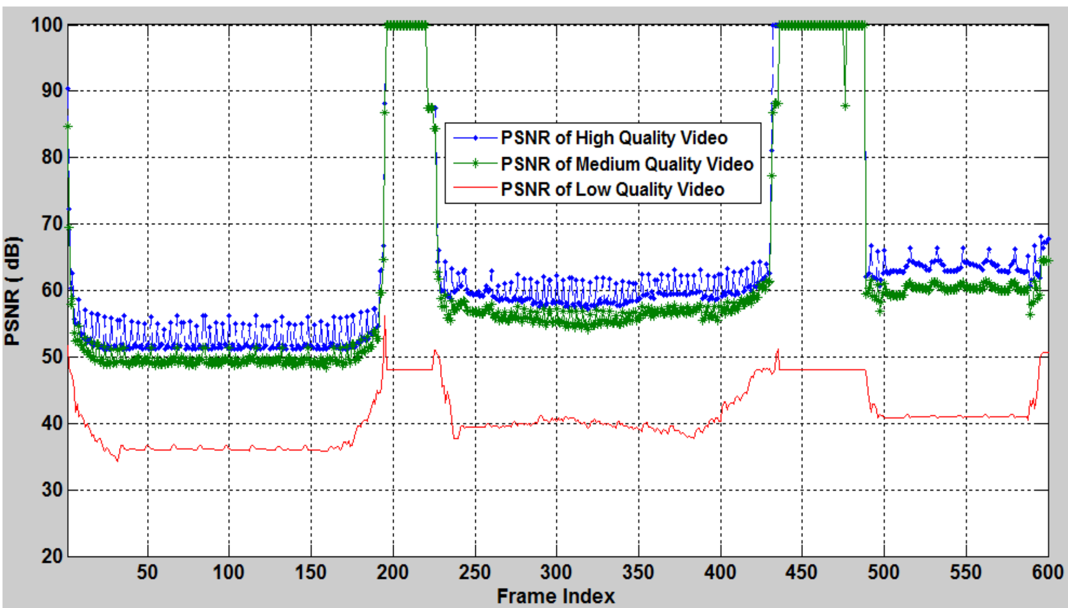

Figure 4.10: PSNR for high, medium and low quality videos

Table 4.5: Summary of results without adaptation

Channel quality	Avg. objective video quality (Q)	Avg. packet loss ratio	Avg. bit rate	Avg. PSNR of input video	PSNR*Q (output video quality)
High	0.9	0.02	3 Mbps	60 dB	54 dB
Medium	0.85	0.08	3 Mbps	60 dB	51 dB
Low	0.4	0.45	3 Mbps	60 dB	24 dB

Table 4.6: Summary of results with adaptation

Channel quality	Avg. objective video quality (Q)	Avg. packet loss ratio	Avg. bit rate	Avg. PSNR of input video	PSNR*Q (output video quality)
High	0.9	0.02	3 Mbps	60 dB	54 dB
Medium	0.93	0.02	1.5 Mbps	55 dB	51.15 dB
Low	0.68	0.25	0.5 Mbps	40 dB	27.2 dB

From results summarized in Table 4.5 and 4.6, we can see that PSNR of input bit stream is almost the same for high and medium quality video. But, average output video quality is improved with considerable reduction in bit rate. PSNR of low quality user is selected above 35 dB to ensure good input video quality based on PSNR to MOS mapping. We can see that approximately 13% video quality gain can be obtained for low channel quality user with adaptation. Savings in average bit rate for low and medium quality videos is also noticeable. Without adaptation, remaining throughput (1.5 Mbps for medium quality user and 2.5 Mbps for low quality user) is a waste of radio spectrum.

Our simulations also show that the bit rate possible for each channel quality of the user is highly dependent on path loss model, multipath model and shadow fading margin used in the simulation. With 10MHz LTE system bandwidth, approximately 50 Mbps data can be sent in the downlink. Considering control plane signals and packet header overheads as 20%, only remaining bandwidth can be allocated for actual video traffic. This corresponds to 40 Mbps. Without proposed GBR selection scheme, same GBR value is used for all the users independent of channel quality. If we are using 3 Mbps GBR for all the users, only 13 users can be accommodated in the cell assuming all the users are accessing same video and no other services are allocated in network. But with CQI based GBR allocation, 3 Mbps is allocated for high quality users, 1.5 Mbps for medium quality users and 0.5 Mbps for low quality users. Considering the equal distribution of users in three channel quality regions, 24 users can be allocated. This helps to admit more number of users and to improve cell coverage.

4.5 Conclusion

The simulation results indicate the advantages of proposed method over video transmission without these adaptations. Packet loss ratio is decreased and objective video quality is increased for all the users in cell coverage area. Increase of video quality for users in cell edge is also highly noticeable. Approximately 13% video quality gain is observed for users at the cell edge using this adaptation scheme. This improved video quality comes with less bit rate video stream and corresponding throughput savings in wired and wireless medium from video server to the users. Furthermore, adaptive GBR selection scheme based on CQI of individual users increase the coverage of the cell considerably. We have done our simulations with static UEs. But, CQI fluctuations can be very high in some of the path loss models and high speed UEs. More frequent adaptations might be required in these scenarios and this will increase the processing load in video server. Overhead bandwidth of the proposed signaling between eNodeB and video server also need to be analyzed with more number of users. These complexities need to be studied further to make it a scalable solution.

Chapter 5

AMC and Scheduling Scheme for Video Transmission in LTE MBSFN Networks

5.1 Introduction

In this chapter, we propose an AMC and scheduling scheme based on MBSFN channel quality measurements to increase video quality of users while maintaining same coverage as single layer video. We further enhance our scheme to increase the spectral efficiency using channel quality distribution information. Unlike existing SVC video transmission schemes over wireless, 3 layers of video are split into 3 sub-channels in the physical layer. This gives more control for physical layer adaptation and scheduling of each video layer separately.

5.2 Proposed Scheme

An AMC and scheduling scheme is proposed for providing graceful degradation of video quality to users in MBSFN area and spectrum savings of MBSFN bearer. SINR values measured at UE terminals are used for adaptation of MCS for base layer and enhancement layers. MCS and scheduling of base layer and enhancement layers are changed based on distribution of users in high, medium and low channel quality regions. This adaptation scheme increases the spectrum efficiency of MBSFN channel and allows serving users with maximum throughput possible in each channel quality region. Three layers of SVC video are split into 3 sub-channels in the physical layer. This gives tight control for physical layer adaptation and scheduling of each video layer separately. UEs are informed about the frequency range of each layer to allow efficient and fast decoding. This scheme ensures coverage of all the users inside MBSFN area and provides graceful degradation of video quality based on channel quality. SINR values are measured in different parts of MBSFN area and the MCS values required for providing coverage in that area are calculated using simulations. We are proposing a scheme to efficiently use the allocated frequency spectrum using adaptive MCS and scheduling scheme. The goal of this scheme is to increase the SE of MBSFN bearer.

The proposed scheme is described below.

Step1: SINR values of users distributed in MBSFN area and MCS value required for providing coverage in that SINR region are pre-calculated using simulations. SE achievable with each of this MCS values are calculated from these simulation results. For a given network, these values need to be pre-calculated.

Step2: Since there are only three video layers, SINR values obtained from Step1 are grouped into three channel quality regions: low, medium and high. This grouping is done based on the modulation scheme used. QPSK, 16 QAM and 64 QAM users are grouped into low, medium and high channel quality regions, respectively in this method. MCS value for each group is selected as the low MCS value in that group to ensure coverage for all users in that group.

Step3: SINR of each UE is measured and informed to MCE of the MBSFN area. MCE calculates the number of users in high, medium and low channel quality regions according to the SINR grouping done in Step2.

Step4: If users are distributed only in high channel quality region, MCS of all three layers of video are set to high.

Step5: If there are users in medium channel quality region, MCS of BL and Enhancement Layer1 (EL1) are set to medium.

Step6: If there are users in low channel quality region, MCS of BL is changed to low for coverage of these users.

Step7: Schedule Enhancement Layer2 (EL2) only if high channel quality user is present. Schedule EL1 only if high or medium channel quality user is present.

Step8: MCS index and frequency range used for SVC quality layers are informed to UEs by MCE.

MCS selection algorithm for BL, EL1and EL2 of the SVC video in MCE is given below as Algorithm 5.1 and decoding procedure in UEs is given below as Algorithm 5.2 below. Channel allocation of multicast channel for different layers is shown in Figure 5.1. F1- F2, F2-F3 and F3-F4 are the frequency ranges allocated for BL, EL1 and EL2, respectively, of SVC video by MCE.

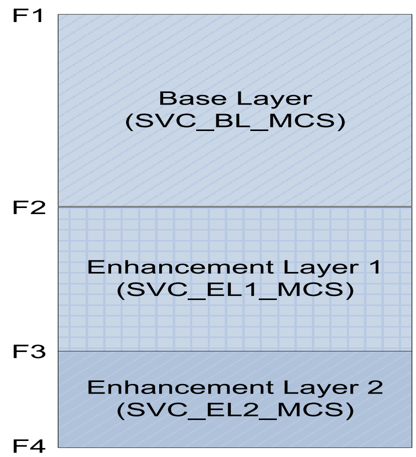

Figure 5.1: Layered transmission of SVC video in MBSFN

Algorithm 5.1. MBSFN dynamic scheduling and AMC algorithm in MCE

1: Measure SINR for users distributed in MBSFN area
2: Calculate MCS value for each SINR value for providing coverage in that specific SINR region
3: Group SINR values to three channel quality regions and MCS values for low, medium and high channel quality regions are measured as explained above in Step2.
4: ***for*** each UE v in MBSFN Area
    calculate SINR(v)
5: ***if*** SINR(v) is member of low channel quality
6: Increment the count of low channel quality users
7: ***else if*** SINR(v) is member of medium channel quality
8: Increment the count of medium channel quality users
9: ***else if*** SINR(v) is a member of high channel quality
10: Increment count of high channel quality users
11: ***end if***
12: ***end for***
13: Set MCS of BL video (SVC_BL_MCS) to high
14: Set MCS of EL1video (SVC_EL1_MCS) to high
15: Set MCS of EL2 video (SVC_EL2_MCS) to high
16: ***if*** Medium channel quality count > 0
17: Set SVC_EL1_MCS to medium
18: Set SVC_BL_MCS to medium
19: ***end if***
20: ***if*** Low channel quality count > 0
21: Set SVC_BL_MCS to low
22: ***end if***
23: ***if*** High channel quality count > 0
24: Schedule BL, EL1 and EL2 for transmission
25: ***else if*** Medium channel quality count > 0
26: Schedule BL and EL1 for transmission
27: ***else if*** Low channel quality count > 0
28: Schedule BL for transmission
29: ***end if***
30: Assign frequency ranges F1, F2, F3, F4 to corresponding SVC layers as shown in Figure 5.1
31: Send F1, F2, F3, F4 and MCS scheme to all UEs in the MBSFN area

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Algorithm 5.2. Pseudo code for decoding procedure in UE

```
if SINR(v) is member of low channel quality
    Decode BL video from F1 to F2 using SVC_BL_MCS
    Final video is BL video
else if SINR(v) is member of medium channel quality
    Decode the BL video from F1 to F2 using SVC_BL_MCS
    Decode EL1 video from F2 to F3 using SVC_EL1_MCS
    Combine BL and EL1videos to produce final video
else if SINR(v) is a member of high channel quality
    Decode the BL video from F1 to F2 using SVC_BL_MCS
    Decode EL1 video from F2 to F3 using SVC_EL1_MCS
    Decode EL2 video from F3 to F4 using SVC_EL2_MCS
    Combine BL, EL1 and EL2 videos to produce final video
end if
```

This scheme is supposed to reduce the spectrum usage in scenarios with no users in low and medium channel quality regions. This dynamic spectrum saving can be used for accommodating other MBSFN bearers or unicast bearers with Best Effort service model. Another option is to increase the bit rate for medium and high quality users in the same MBSFN bearer to obtain higher video quality. This scheme can be implemented with minimal changes in MCE and eNodeB nodes of LTE networks. Sub frames used for MBSFN transmission in MBSFN area are marked as MBSFN sub frame by MCE. At the beginning of each MCH Scheduling Period, a MAC control signal is transmitted to convey MCH Scheduling Information (MSI). For dynamic scheduling and adaptation change, MSI should be set as a low value in the MBSFN area. Further, this scheme can be adapted to multicast and broadcast services in other OFDMA based wireless networks such as WiMAX Multicast Broadcast Service (MCBCS) single frequency operation [14].

5.3 Simulation Model

MBSFN area is created with 7 cells organized as shown in Figure 5.2. UE measurements are done on entire cell coverage area of eNodeB_1, and other surrounding cells assist eNodeB_1 in MBSFN operation. All eNodeBs transmit same data in specific MBSFN bearer frequency range to users in their cell area. Infinite bandwidth model is used in the core network links and routers. SVC video traces are created using JSVM SVC encoding software [26]. SVC encoder setting used is shown in Table 5.1. LTE system configuration used in the simulation is given in Table 5.2.

Table 5.1: SVC encoder settings

Parameters	Value
Number of Frames	500
Resolution	QCIF Base Layer and CIF Enhancement layer
Frame Rate	30 fps
GOP	16
Encoding Bit Rate	VBR
SVC scalability layers	2 Spatial layers and 3 Quality layers.

Table 5.2: LTE system parameters

Parameters	Value
LTE Mode	FDD
Traffic direction	Downlink
eNodeB operating power	46 dBm
Inter Site Distance (ISD)	1732 meter
Number of OFDM symbols per subframe	12
Cyclic prefix	16.6 us
Carrier frequency	2.5 GHz
LTE bandwidth	20 MHz (100 PRBs)
User location	Users with different Channel qualities (high, medium and low) equally distributed in the MBSFN area
Path loss model	Suburban macrocell
Multipath channel model	ITU Pedestrian A
# of Tx Antennas	1
# of Rx Antennas	1
RLC mode	UM
Maximum number of HARQ retransmissions	3

Users in the MBSFN area is divided into 3 channel quality regions: High, Medium and Low. Simulations are done with SVC trace of Football video [44] generated using JSVM. Spatially scalable video with QCIF base layer, and CIF enhancement layer is used for the simulations.

Since multicast group can contain heterogeneous receivers, this is a valid requirement in the real-time scenario. One base quality layer with data rate of 256 kbps, first enhancement layer with 150 kbps and second enhancement layer with 100 kbps are encoded for multicast transmission. Simulations are run for 3 times and average results are used for calculation.

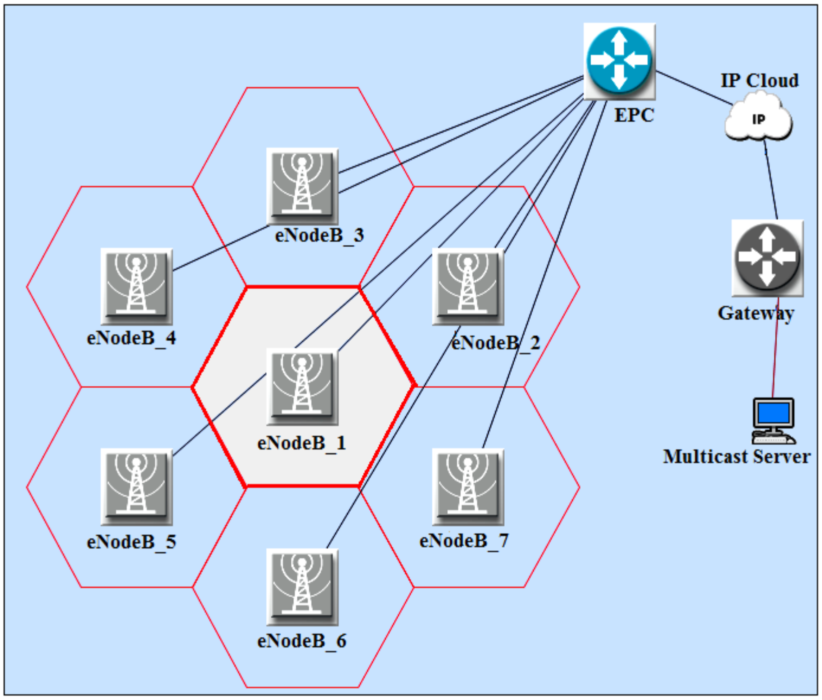

Figure 5.2: MBSFN Network diagram used for simulation

5.4 Simulation Results and Analysis

SINR values of users distributed in MBSFN area and MCS value required for providing coverage in that SINR region are measured using simulations. SE achievable with each of this MCS values are also calculated from simulation results. SINR values and corresponding MCS and SE values measured are shown below in Table 5.3. SINR values are grouped to low, medium and high channel quality regions, and low MCS in each group is selected as MCS value for that specific group. These groups, SINR range, and MCS value selected are shown in Table 5.4.

Frequency spectrum and number of PRBs used by each video layer is calculated and total frequency spectrum utilization is compared for three different channel quality distribution scenarios. One PRB corresponds to 180 Khz of frequency spectrum, and it is the smallest possible allocation block in the frequency domain. MCS is changed for base and enhancement layers based on the channel quality distribution and total spectrum usage is calculated in each scenario. Spectrum savings in Scenario 2 and Scenario 3 are calculated and compared with Scenario1 (existing approach).

Table 5.3: SINR, MCS and SE mapping in MBSFN simulation

SINR (dB)	MCS Index	Modulation Scheme	SE of MBSFN bearer (bits/second/Hz)
-3.85	1	QPSK	0.02009
-2.1	3	QPSK	0.03231
-0.35	5	QPSK	0.05157
1.4	7	QPSK	0.07517
3.15	9	QPSK	0.10078
4.9	12	16QAM	0.12657
6.65	14	16QAM	0.16407
8.4	16	16QAM	0.20625
10.15	19	64QAM	0.23404
11.9	21	64QAM	0.28477
13.65	23	64QAM	0.33448
15.4	25	64QAM	0.38772
17.15	27	64QAM	0.43845
18.9	28	64QAM	0.47612

Table 5.4: Channel quality regions, MCS and spectral efficiency

Channel quality regions	SINR value(dB)	MCS	Spectral efficiency of MBMS bearer (bps/Hz)
High	10.15 - 18.9	19	0.23404
Medium	4.9 - 8.4	12	0.12657
Low	-3.85 - 3.15	1	0.02009

A. Scenario 1 (Existing Approach)

Users are distributed in all channel quality regions in this scenario. This corresponds to previous work on single cell multicast cellular systems with static AMC selection for video layers [10, 16]. In this case base layer is sent with MCS 1, EL 1 is sent with MCS 12, and EL 2 is sent with MCS 19. These MCS values are selected based on measurements shown in Tables 5.3 and 5.4. Spectrum utilization in number of PRBs and Hz is given below in Table 5.5. 71 PRBs are used for base layer video and total spectrum usage is 14.58 MHz in each MBSFN radio subframe.

Table 5.5: Spectrum usage with users distributed in all channel quality regions

SVC video layer	Data rate (kbps)	MCS used	Number of PRBs used	Spectrum Used (MHz)
Base	256	1	71	12.78
Enhancement 1	150	12	7	1.26
Enhancement 2	100	19	3	0.54

B. Scenario 2

Users are distributed across high and medium channel quality regions in this scenario. In this case base layer is sent with MCS 12, EL 1 is sent with MCS 12, and EL 2 is sent with MCS 19. Spectrum utilization in number of PRBs is given below in Table 5.6.

Table 5.6: Spectrum usage with users distributed only in medium and high channel quality regions

SVC video layer	Data rate (kbps)	MCS used	Number of PRBs used	Spectrum Used (MHz)
Base	256	12	12	2.16
Enhancement 1	150	12	7	1.26
Enhancement 2	100	19	3	0.54

Only 12 PRBs are used for base layer video with MCS changed to 12. Total spectrum usage is 3.96 MHz in each MBSFN radio sub frame.

C. Scenario 3

Users are distributed only in high channel quality regions in this scenario. In this case, all video layers are sent with MCS 19. Spectrum utilization in number of PRBs is given below in Table 5.7. Total spectrum usage is 2.52 MHz in each MBSFN radio subframe. Since high MCS is used for all the layers in this scenario, spectrum savings compared to Scenario 1 is very high in this case. Savings in radio spectrum for all three scenarios are compared in Table 5.8.

Table 5.7: Spectrum usage with users distributed only in high channel quality regions

SVC video layer	Data rate (kbps)	MCS used	Number of PRBs used	Spectrum Used (MHz)
Base	256	19	7	1.26
Enhancement 1	150	19	4	0.72
Enhancement 2	100	19	3	0.54

Simulation results for the proposed scheme in Scenario 2 and 3 indicate advantages of using user distribution information to optimize MCS selection for different layers, compared to MCS selection schemes presented in references [10, 16]. Savings of radio spectrum in the order of 72 to 82% is observed with the proposed method in different channel quality distributions. Even though Scenario 3 is less probable in real-time networks, there is a good probability for users distributed only in high quality and medium quality regions in an MBSFN network (Scenario2), and a considerable amount of spectrum savings is observed for this case.

Table 5.8: Percentage savings in spectrum with proposed method

Scenario	Total frequency spectrum used without proposed method (MHz)	Total frequency spectrum used with proposed method (MHz)	Savings in radio spectrum
Scenario1	*14.58*	*14.58*	-
Scenario 2	14.58	3.96	72.83 %
Scenario 3	14.58	2.52	82.71 %

5.5 Conclusion

A dynamic AMC and scheduling scheme for MBSFN based SVC video transmission in LTE network is proposed in this chapter. Modulation and coding of different quality layers of SVC encoded video is changed dynamically based on the distribution of users in the coverage area. Savings of radio spectrum in the order of 72 to 82% is observed in different user distribution scenarios with the proposed method. This spectrum can be used for serving other MBSFN, single cell MBMS or unicast bearers or can be used for increasing the video quality of same MBSFN bearer. The results discussed are simulated with static user distribution, hence the control channel communication overhead during the user mobility is not considered. Further work can be done to improve the current scheme by considering the user mobility without adding much overhead to the existing system.

Chapter 6

Conclusions and Future Work

6.1 Conclusions

Adaptations and scheduling methods for video streaming over LTE networks using SVC is presented in this thesis. The advantages of proposed methods in multicast and unicast scenarios are obvious from the analysis and simulation results presented in the above chapters. The simulation results indicate the advantages of proposed method over video transmission without these adaptations. Different parameters like throughput, packet loss ratio, delay, and jitter are compared with H.264 single layer video for unicast and multicast scenarios using different kinds of scalabilities. Results show that considerable packet loss reduction and throughput savings (18 to 30%) with acceptable video quality are achieved with adaptation scheme based on SVC compared to H.264.

Packet loss ratio is decreased and objective video quality is increased for all the users in cell coverage area for proposed adaptation method for unicast based on channel quality information. Increase of video quality for users in cell edge is also highly noticeable. Approximately 13% video quality gain is observed for users at the cell edge using this adaptation scheme. This improved video quality comes with less bit rate video stream and corresponding throughput savings in wired and wireless medium from video server to the users. Furthermore, adaptive GBR selection scheme based on CQI of individual users increase the coverage of the cell considerably. In the proposed MBSFN video streaming scheme, modulation and coding of different quality layers of SVC encoded video is changed dynamically based on the distribution of users in the coverage area. Savings of radio spectrum in the order of 72 to 82% is observed in different user distribution scenarios with the proposed method. This spectrum can be used for serving other MBSFN, single cell MBMS or unicast bearers or can be used for increasing the video quality of same MBSFN bearer.

We have simulated and presented the results for LTE networks, but proposed schemes can be implemented in other OFDMA based 4G networks like WiMAX with minor changes.

6.2 Future Work

Cross layer adaptations and scheduling proposed in this thesis are focused on the channel quality information of the users. However, adaptations using other network parameters such as congestion, delay and end user parameters such as mobile processing power, battery power are also possible with SVC. Adaptations based on the video content itself are possible and need to be explored further. Our work was specific to video transmission in the downlink direction. This can be expanded to accommodate the adaptations possible for uplink video transmission in 4G networks. Sending real time videos through a simulator and testing the visual quality differences with different types and mix of scalabilities will give more insight about this interesting research area. MBSFN and other advanced features of LTE make attractive options for optimizing the bandwidth utilization and providing enhanced QoS to cellular users. Further work is planned in the direction of studying cross layer design options possible in video streaming over LTE-Advanced networks.

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