

https://dspace.org/

Scheduling in digital video broadcasting second generation terrestrial (DVB-T2) systems

Khalanyane, Reitumetse Nkali Charlotte

2018

https://hdl.handle.net/10539/27313

Khalanyane, Reitumetse Nkali Charlotte (2018) Scheduling in digital video broadcasting second generation terrestrial (DVB-T2) systems, University of the Witwatersrand, Johannesburg, https://hdl.handle.net/10539/27313

Downloaded from a DSpace repository. https://dspace.org/

SCHEDULING IN DIGITAL VIDEO BROADCASTING SECOND GENERATION TERRESTRIAL (DVB-T2) SYSTEMS

By

Reitumetse Nkali Charlotte Khalanyane

A dissertation submitted in fulfilment to the requirements for the degree of Master of Science in Engineering in the School of Electrical and Information Engineering at the University of the Witwatersrand, Johannesburg, South Africa.

Supervisor: Prof. Fambirai Takawira

Co-Supervisor: Dr. Olutayo Oyerinde

September 2018

Declaration

I, Reitumetse Nkali Charlotte Khalanyane, declare that this dissertation is my own original work unless otherwise acknowledged or referenced. It is submitted in fulfillment of the award of the Degree of Master of Science in Engineering, in the School of Electrical and Information Engineering, Faculty of Engineering and Built Environment, University of the Witwatersrand. It has not been submitted for qualification at any other academic institution.

Signed: _____

_____ Day of_____ year_____

Dedication

In the name of the Father, the Son and the Holy Spirit I dedicate this dissertation to my parents, Tankie and ‘Mahlaoli Khalanyane, for their love, encouragement, support, patience and sacrifices over the years.

Acknowledgements

Thanks to Almighty God for giving me the strength, ability and wisdom to understand, learn and complete this study.

I would like to express my sincere gratitude to my supervisor Prof. Fambirai Takawira and co-supervisor Dr. Olutayo Oyerinde whose expertise, understanding, generous guidance and support made it possible for me to complete this research. I truly appreciate all the efforts and contributions made.

I am hugely indebted to Dr. Stephen Chabalala for finding time in his schedule to assist me, for showing interest in my research and making helpful comments and contributions regarding the topic of my research.

I am thoroughly grateful to my parents, Mr. Tankie and Mrs. Mahlaoli Khalanyane, for their patience as well as emotional and financial support throughout this study. I would also like to express my gratitude to my siblings Kekeletso, Thatohatsi, Resetse and Teliso for their support and encouragement. I am thankful to my nieces and nephew Bonang, Phoka, Bonolo and Umphile for being a source of motivation. I am exceptionally grateful to Hlalele Heisi for his love and support throughout my study.

I would like to express a great thanks to my friends and colleagues, particularly Elizabeth Haikali, Molefi Makuebu, Lerato Lesenyeho, Tankiso Lekhooa, Liakae Mohlotsane, Mats'olo Seloanayane, Oluwaseun Erinle and Alex for their support and encouragements. I would like to pass a special thank you to the Holy Trinity Catholic church student choir and the church as a whole for being my home away from home and uplifting me spiritually.

Last but not least, I acknowledge with sincere appreciation, the financial support of SENTECH and Wits University Postgraduate Merit Award (PMA).

Abstract

The ever-increasing need for high quality video and new video content in digital video broadcasting comes with a demand for more bandwidth. However, bandwidth is already a scarce resource and the challenge faced is how to satisfy this demand in the limited bandwidth. The most recent digital terrestrial television (DTT) standards and video compression algorithms attempt to use the spectrum more efficiently but this is still not enough. Proper resource management procedure, particularly scheduling, are therefore essential in ensuring that the spectrum is used even more efficiently. The objective of this research is to design and implement resource allocation solutions for Digital Video Broadcasting Second Generation Terrestrial (DVB-T2) in order to use resources more efficiently and improve the system performance while considering quality of service requirements (QoS). Another problem in DVB-T2 is that resource allocation is not standardized and this comes with disadvantages such as lack of interoperability between hardware from different vendors which limits customers to single vendor. This research therefore proposes standardization of scheduling for DVB-T2.

The research addresses resource allocation in DVB-T2 at three levels. Firstly, physical layer pipes (PLP) scheduling schemes namely dynamic statistical multiplexing (D-StatMux), modified largest weighted buffer occupancy first (MLWBOF) and exponential proportional fair (EXP/PF) are presented and analyzed. Secondly, a service scheduling scheme for scalable videos is also presented in order to exploit the bit rate adaptability of scalable video coding (SVC). Lastly, an existing time slice allocation scheme is adopted in this study for timely delivery of services. The schedulers are implemented for three scenarios: PLP scheduling with increasing number of services, Service and PLP scheduling with over-packing and Service and PLP Scheduling with time slice allocation with over-packing. Results obtained from carrying out simulations show that the proposed schedulers give better overall performance than existing solutions.

On this basis, it is recommended that scheduling be standardized for DVB-T2 and the proposed scheduling schemes should be considered.

Table of Contents

Declaration i

Dedication ii

Acknowledgements iii

Abstract iv

Table of Contents v

List of Tables viii

List of Figures ix

List of Abbreviations xi

List of Notations xiv

Chapter 1 Introduction 1

1.1 Digital Terrestrial Television 1

1.1.1 Source Coding and Compression 2

1.1.2 Service Multiplexing and Transport 3

1.1.3 Physical layer 3

1.2 Recent DTT Standards 5

1.2.1 DVB-T2 5

1.2.2 ATSC 7

1.2.3 ISDB-T 9

1.2.4 DTMB 10

1.3 Problem Statement 11

1.4 Research Question 13

1.5 Research Methodology 13

1.6 List of Publications 14

1.7 Organization of the Dissertation 14

Chapter 2 DVB-T2 System Components 16

Table of Contents

2.1 Introduction 16

2.2 DVB-T2 16

2.2.1 System Architecture 16

2.2.2 Capacity and Bit rates 27

2.2.3 Bandwidth Allocation 30

2.2.4 Time Slice Allocation 31

2.3 SVC and Resource Mapping 32

2.3.1 Scalable Video Coding 32

2.3.2 Types of Scalability 34

2.3.3 Resource Mapping – Related Work 38

2.4 Chapter Summary 39

Chapter 3 PLP scheduling 40

3.1 Introduction 40

3.2 System Model 40

3.2.1 System Model Description 40

3.2.2 System Constraints 41

3.3 PLP Scheduling Algorithms 42

3.3.1 D-StatMux 42

3.3.2 MLWBOF 45

3.3.3 EXP/PF 47

3.4 Simulations 49

3.4.1 Simulation Model 49

3.4.2 Performance Measures 54

3.4.3 Algorithm for Developing Results 56

3.4.4 Simulation Results and Discussion 57

3.5 Chapter Summary 61

Chapter 4 Joint PLP and Service Scheduling 63

Table of Contents

4.1 Introduction 63
4.2 System Model 64
4.2.1 System Model Description 64
4.2.2 System Constraints 65
4.3 Service Scheduling 66
4.3.1 Base Layer Allocation 67
4.3.2 Enhancement Layer Allocation 68
4.4 Computational Analysis 69
4.5 Simulations 70
4.6 Chapter Summary 73
Chapter 5 Joint Service and PLP scheduling with Time Slice Allocation 75
5.1 Introduction 75
5.2 System Model 75
5.2.1 System Model Description 75
5.2.2 Time Slice Allocation Constraints 77
5.3 Time Slice Allocation 77
5.4 Simulations 81
5.5 Chapter Summary 84
Chapter 6 Conclusions and Future Work 86
6.1 Conclusions 86
6.2 Future Research 87
References 89

List of Tables

Table 1.1 A Comparison between DVB-T and DVB-T2 7
Table 2.1 BB Frame size in bits.....19
Table 2.2 Bit-Mapping into Constellation 20
Table 3.1 DVB-T2 System Parameters.....49
Table 3.2 Input Services Bit Rates and Transmission Delays 51
Table 3.3 Bit Rates of Tokyo Olympics Video Layers.....51
Table 3.4 Bit Rates of BBC News Video Layers.....51
Table 3.5 Bit Rates of Terminator Video Layers.....52
Table 3.5 Algorithm for Development of Results 56
Table 4.1 Base layer Allocation Algorithm 68
Table 4.2 Enhancement layer Allocation Algorithm 69

List of Figures

Figure 1.1 ITU DTTB Model....2
Figure 1.2 Illustration of OFDM Modulation Scheme....4
Figure 1.3 Illustration of VSB Modulation Scheme....5
Figure 1.4 Channel Bonding Illustration....9
Figure 1.5 BST-OFDM Modulation Scheme....10
Figure 1.6 TDS-OFDM Frame Structure....11
Figure 2.1 DVB-T2 system block Diagram....17
Figure 2.2 DVB-T2 Input Processing Module for Input Mode A....19
Figure 2.3 Mode Adaptation for Input Mode B....19
Figure 2.4 Stream Adaptation for Input Mode B....20
Figure 2.5 Baseband and FEC Frames Structure....21
Figure 2.6 16 QAM Constellation....22
Figure 2.7 A Rotated 16 QAM Constellation....22
Figure 2.8 Time Interleaving Options....24
Figure 2.9 DVB-T2 Framing Structure....25
Figure 2.10 Scattered Pilot Patterns Examples....27
Figure 2.11 Cell Mapping in DVB-T2....28
Figure 2.12 Illustration of Simulcasting....33
Figure 2.13 Illustration of Scalable video Coding....34
Figure 2.14 An Example of Temporal Scalability....36
Figure 2.15 An Example of Spatial Scalability....37
Figure 2.16 An Example of Quality Scalability....37
Figure 3.1 System Model for PLP Scheduling....41
Figure 3.2 Frame Distribution Graphs for Sport Trace Files....52
Figure 3.3 Frame Distribution Graphs for News Trace Files....53
Figure 3.4 Frame Distribution Graphs for Movie Trace Files....54
Figure 3.5 System Throughput with increasing number of services....59
Figure 3.6 BB Frame loss with increasing number of services....60
Figure 3.7 Fairness with increasing number of services....60
Figure 4.1 System Model for Joint Service and PLP Scheduling....64
Figure 4.2 System Throughput with increasing number of services....71

List of Figures

Figure 4.3 BB Frame loss with increasing number of services 72
Figure 4.4 Fairness with increasing number of services 73
Figure 5.1 System Model for Joint Service and PLP Scheduling with Time Slice Allocation 76
Figure 5.2 Slice Allocation Scheme's proposed data slices in a T2 Frame 77
Figure 5.3 Addressing Scheme 79
Figure 5.4 Slice Allocation Tree Illustration 80
Figure 5.5 System Throughput with over-packing for Joint Service and PLP Scheduling with Time Slice Allocation 83
Figure 5.6 System BB Frame Loss due to Time Slice Allocation with over-packing for Joint Service and PLP Scheduling with Time Slice Allocation 84

List of Abbreviations

APSK	Amplitude Phase Shift Keying
ATSC	Advanced Television Systems Committee
AVC	Advanced Video Coding
BB	Baseband
BCH	Bose-Chaudhuri-Hocquenghem
BICM	Bit Interleaved Coding and Modulation
BST-OFDM	Band Segmented Transmission OFDM
CBR	Constant Bit Rate
CGS	Coarse Grain Scalability
C-OFDM	Coded OFDM
D- StatMux	Dynamic Statistical Multiplexing
DTMB	Digital Terrestrial Multimedia Broadcast
DTMB-A	Digital Terrestrial Multimedia Broadcast Advanced
DTTB	Digital Terrestrial Television Broadcasting
DVB-H	Digital Video Broadcasting for Handhelds
DVB-T	Digital Video Broadcasting Terrestrial
DVB-T2	Digital Video Broadcasting Second Generation Terrestrial
EXP/PF	Exponential Proportional Fairness
FEC	Forward Error Correction
FEF	Future Extension Frame
FFC	Federal Communications Commission
FFT	Fast Fourier Transform
FGS	Fine Grain Scalability
GPS	Generalized Processor Sharing
GS	Generic Stream
HDTV	High Definition Television
HEVC	High Efficiency Video Coding

IF	Interleaving Frame
IP	Internet Protocol
IPL	Inter-Layer Prediction
ISDB-T	Integrated Services Digital Broadcasting – Terrestrial
LDM	Layered Division Multiplexing
LDPC	Low Density Parity Check
LTE	Long-Term Evolution
MGS	Medium Grain Scalability
MLWBOF	Modified Largest Weighted Buffer Occupancy First
MLWDF	Modified Largest Weighted Delay First
MPEG	Moving Picture Expert Group
NAL	Network Abstraction Layer
OFDM	Orthogonal Frequency Division Multiplexing
PAPR	Peak to Average Power Ratio
PF	Proportional Fair
PLP	Physical Layer Pipes
PN	Pseudo Noise
PSK	Phase Shift Keying
PSNR	Peak Noise to Signal Ratio
QAM	Quadrature Amplitude Modulation
QoE	Quality of Experience
QoS	Quality of Service
RM	Resource Management
RS	Reed Solomon
RT	Real-time
SFN	Single Frequency Network
SNR	Signal to Noise Ratio
StatMux	Statistical Multiplexing
SVC	Scalable Video Coding

TDS-OFDM	Time Domain Synchronous OFDM
TFS	Time Frequency Slicing
TIB	Time Interleaving Block
TS	Transport Stream
UHD	Ultra High Definition
VBR	Variable Bit Rate
VSB	Vestigial Sideband

List of Notations

B_{T2}	Total number of BB frames in a T2 frame
B_{TFS}	Total number of BB frames in a TFS frame
B_{cells}	Total number of cells in a TFS frame
$B_{discarded}$	Total size of discarded BB frames of PLP i over the duration of a T2 frame
B_i	Buffer occupancy of PLP i
B_l	FEC block length
B_{max}	Buffer size
B_{size}	Total size of all the BB frames arriving at the service buffer of PLP i over the duration of a T2 frame
C_{FC}	Number of data cells in a frame closing symbol
$C_{channel}$	Channel capacity
C_{data}	Number of data cells in a data symbol
C_{input}	Input capacity which is the total allowable input bit rate
C_{p2}	Number of data cells in a P2symbol
C_{tot}	Total number of cells in a T2 frame
D_{HOL}	Head of line delay
D_{L1}	Number of cells carrying L1 signalling information
D_{PLP}	Number of cells available for transmission of PLPs
D_i	Delay constraint of PLP i
F_R	Frame Rate

F_p	Frame period
I_s	Data slice interval
$K_{bch,i}$	BCH coding length of PLP i
K_{bch}	Number of BCH encoder input bits
$L_{BBframes}$	BB frame loss ratio
L_{data}	Number of data symbols
L_g	Number of quality layers of SVC service g
$N_{B,i}(R_i(t))$	Number of BB frames required by PLP i to achieve the instantaneous bit rate $R_i(t)$
N_{B_max}	Number of FEC blocks per T2 frame
N_{T2}	Number of T2 frames
N_{bch}	Number of BCH encoder output bits
N_{cells}	Number of data cells per FEC block
N_{node}	Number of nodes in a binary tree
N_{p2}	Number of P2 symbols
N_s	Number of time slices in a T2 frame
$R_{g,l}$	Average bit rate of the l^{th} layer of an SVC service g
R_g	Average bit rate of SVC service g
$R_i(t)$	Instantaneous bit rate of PLP i at time t
$R_i^{av}(t)$	Average throughput of PLP i at time t
$R_{throughput}$	Actual average throughput requirement of PLP i over the duration of a T2 frame
S_i	Number of slices required by PLP i

$S_{throughput,i}$	Average throughput of PLP i over the duration of the simulation
T_{P1}	Duration of a P1 symbol
T_f	Duration of a T2 frame
T_{fs}	Duration of a super frame
T_i	Last cell address of PLP i
T_s	Duration of a data symbol
R^{av}	Average system throughput over the duration of the simulation
W_i	Weight associated with PLP i
X_p	Total size of in bytes of encoded video frames for a given frame period p
X_p^w	Size of in bytes of encoded video frames of video layer w for a given frame period p
$a_{l,c}$	Address of the data cell on data symbol l and cell index c of the symbol
b_i	Number of BB frames allocate to PLP i
$b_i(t)$	Number of BB frames allocated to PLP i at time t
c_i	Code rata of PLP i
l_g	Index of a video layer of an SVC service g
l_{node}	Level of a node in a slice allocation tree
m_i	Modulation index of PLP i
$r_{g,l}$	Half the peak bit rate of the l^{th} layer of an SVC service g
r_g	Peak bit rate of an SVC service g
r_i	Data rate of PLP i

$r_{i\ j}$	Throughput of PLP i on the j^{th} T2 frame
$r_i(t)$	Data rate achieved by PLP i at time t
t_c	Time window over which fairness is imposed
$t_j(i)$	Address of the jth data slice PLP i is mapped to
$t_{start}(i)$	Address of the first data slice PLP i is mapped to
δ_i	Allowable packet/BB frame loss due to buffer overflow for PLP i
σ_i	Allowable loss due to packet delay for user i
h	Height of a complete binary tree
F	Fairness index
G	Set of SVC services
L	Height of a slice allocation tree
L	Height of slice allocation tree
M	Number of PLPs allocated BB frames
N	Number of configurable PLPs in the T2 system
Q	Number of PLPs allocated time slices
R	Total data rate of the system
τ	Delay threshold

Chapter 1 Introduction

1.1 Digital Terrestrial Television

Digital Terrestrial Television (DTT) was introduced as a move from analogue transmission to digital transmission. The transition was influenced by the advances in digital technology in both algorithmic and hardware implementations, and its capability of improving spectrum efficiency and robustness against propagation effects through error correction codes[1]. The advantages of DTT are similar to those of digital versus analogue platforms in all telecommunication system. The advantages include efficient use of spectrum as more content can be transmitted in one channel, increased transmission capacity hence support for new services such HDTV and multimedia or interactive services, support for mobile and portable reception and improved reception and picture quality because DTT is resistant to channel distortions (multi-path, noise and interference). Migrating from analogue to digital allows flexible and efficient use of the spectrum through single frequency networks (SFNs). In a SFN two or more transmitters operate on the same frequency without causing interference hence improving the spectral efficiency of the system. The migration means part of the spectrum that was used for analogue can be released for other uses such as mobile communication. DTT systems were designed to allow large coverage for fixed rooftop antennas and as large as possible reception for portable indoor reception. Mobile reception can also be achieved in some standards. DTT resulted in various standards across the world. The existing standards are:

a) Advanced Television System Committee (ATSC) , an American standard that has been adopted in a number of countries including USA, Canada, Mexico, Honduras, El Salvador and South Korea;
b) Digital Video Broadcasting-Terrestrial (DVB-T) which is a European standard and has been adopted in the whole of Europe, some parts of Africa including Lesotho and South Africa, and Australia. Many variants for fixed, portable and mobile reception exist in this family;
c) Integrated Service Digital Broadcasting- Terrestrial (ISDB-T) developed in Japan and used in Japan and Brazil;

d) Digital Terrestrial Multimedia broadcasting (DTMB), a Chinese standard. It has been adopted in China, Hong Kong, Cuba and Macau.

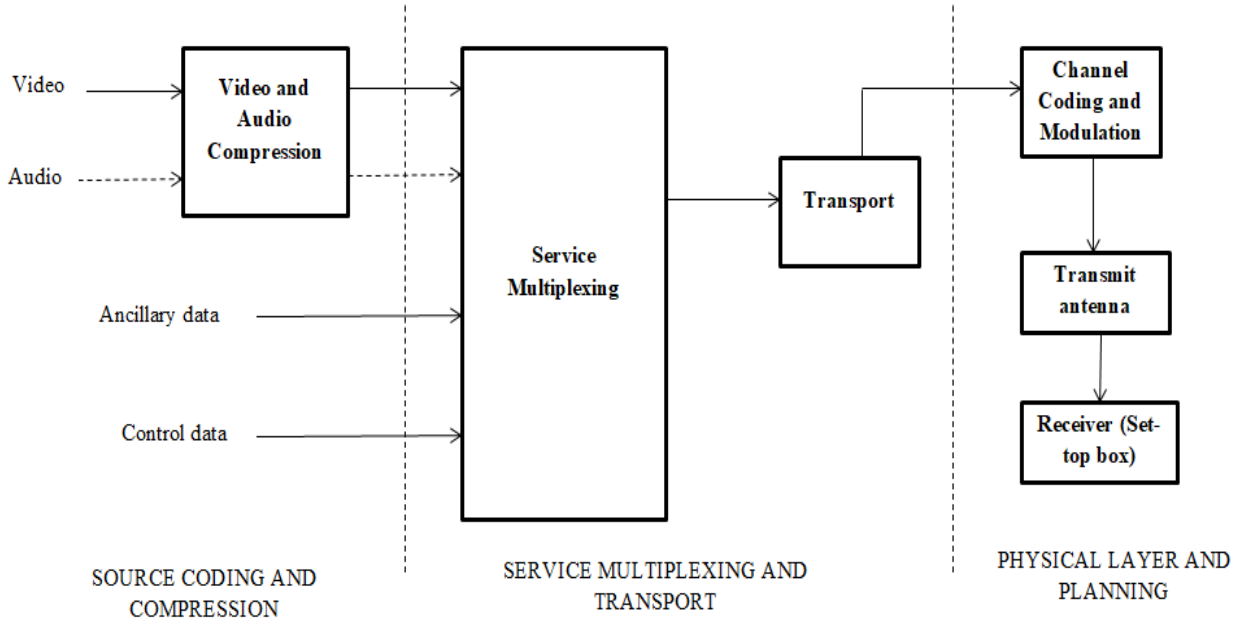


Figure 1.1 ITU DTTB Model

The DTT model consists of four subsystems (Figure 1.1) : source coding and compression, service multiplexing and transport, physical layer and planning [2].

1.1.1 Source Coding and Compression

Source coding and compression refers to data compression mechanisms designed to reduce the data rate of large streams of data. This is essential in DTT because it reduces the amount of bandwidth needed to transmit data streams. Proper compression techniques must minimise the number of bits needed to represent the information but it should be able to recreate the information precisely. Any losses should be transparent to human perception. The Moving Picture Expert Group (MPEG) was formed to standardise video and audio compression techniques for digital storage [1]. Earlier DTT standards published in the 1990s used MPEG-2 for video and audio coding. As technology advances source coding and compression has become more efficient with lower implementation complexity. MPEG-4 was designed to improve on the prior standards. Later standards have also used MPEG-4 part 10 also known as H.264/AVC especially for delivering HDTV. The scalable extension of H.264/AVC, H.264/SVC, has been adopted as one of the video codecs for DVB broadcast services [3].

The most recent standards ATSC.3.0 and DTMB-Advanced (DTMB-A) have adopted H.265/HEVC and its scalable extension as one of the video codecs for video broadcasting[2].

1.1.2 Service Multiplexing and Transport

Service multiplexing and transport divides streams of each service into packets containing unique identifiers for each packet and packet type then multiplexes those service packets into a single data stream and finally combines particular service data streams into a single broadcast channel for simultaneous transmission. The most popular stream structure is the MPEG-2 audio and video Transport Stream (TS) structure. All published DTT standards use MPEG TS. DVB-T2 introduces some other stream structures referred to as Generic Streams (GS). Rather than using MPEG-TS for streaming and file delivery, ATSC 3.0 uses Internet Protocol (IP) encapsulation. Multiplexing should take into consideration appropriate transport mechanisms which provide interoperability between digital media such as terrestrial, satellite and cable broadcasting, computer interfaces, recording devices and receivers [2].

1.1.3 Physical layer

The physical layer contains channel coding and modulation. Channel coding entails adding extra bits to the compressed data in order to detect and correct errors. Most DTT standards use Reed Solomon (RS) and convolution codes for forward error coding (FEC). DVB-T2, ATSC 3.0 and DTMB-A use Bose-Chaudhuri-Hocquenghem (BCH) and low density parity check (LDPC) codes which are capable of attaining improved system performance compared to RS and convolution codes [4]. Modulation scheme converts the error coded stream into modulated signal on one or more carriers. In terrestrial broadcasting where the channel experiences severe fading in time and frequency, sophisticated modulation is required. Two approached have been documented in DTT: coded OFDM (COFDM) and 8-level vestigial sideband (8-VSB) modulation. OFDM is a multicarrier modulation scheme that splits high data rate data streams into closely spaced orthogonal subcarriers that are transmitted in parallel and each subcarrier can be modulated with a digital modulation scheme such as PSK, QAM, APSK etc., at low symbol rate. Although the sidebands overlap, they can still be received without interference because they are orthogonal to each other. Figure 1.2 shows a diagrammatic presentation of OFDM. VSB encodes data by varying amplitude of a single carrier frequency and removing portions of one of the redundant sidebands forming a vestigial sideband signal as shown in Figure 1.3. 8-VSB includes 8 amplitude levels. All terrestrial standards use OFDM except ATSC 1.0 which uses VSB for compatibility with

older NTSC schemes. An advantage of OFDM is that it is less susceptible to interference but thanks to advances in equaliser technology VSB is able to attain the same comparable performance to OFDM in multipath conditions [1]. However, OFDM is capable of supporting SFNs which is not possible with VSB. VSB on the other hand uses a single side-band for transmission making it more bandwidth efficient than OFDM.

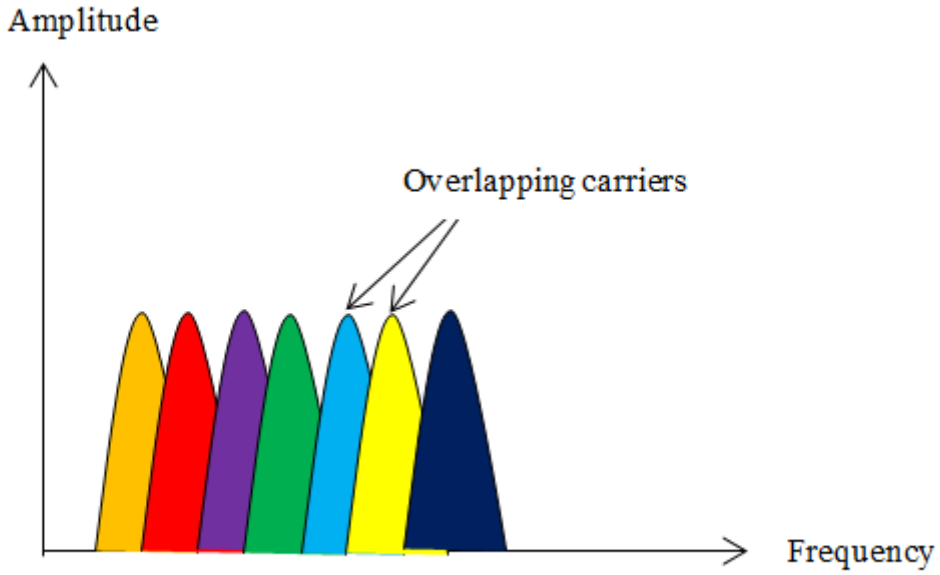


Figure 1.2 Illustration of OFDM Modulation Scheme

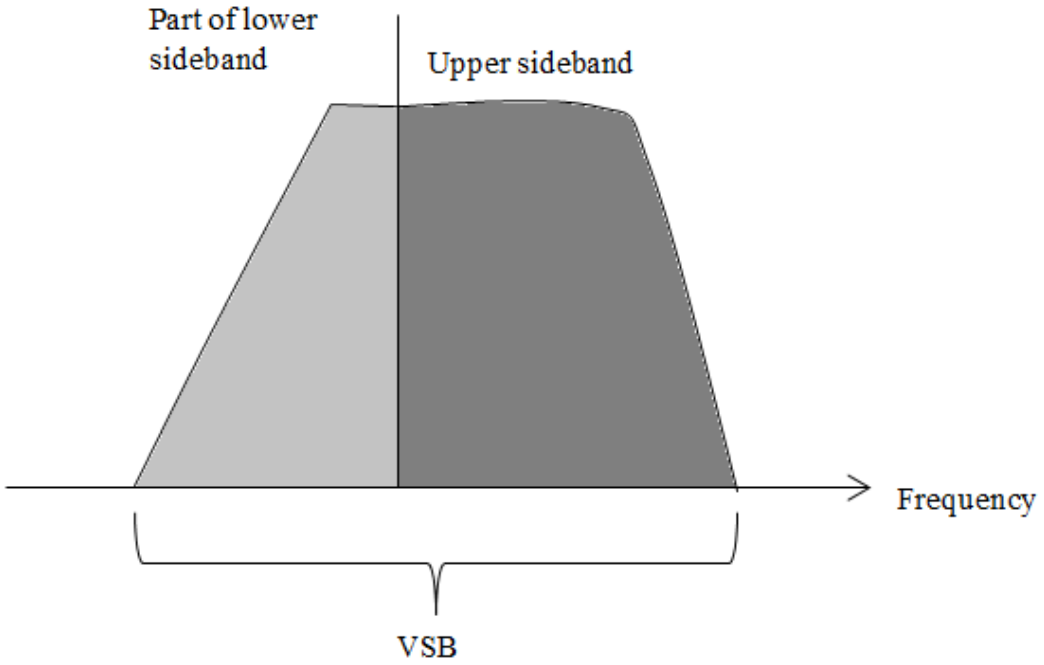


Figure 1.3 Illustration of VSB Modulation Scheme

1.2 Recent DTT Standards

1.2.1 DVB-T2

The first DVB standard for terrestrial broadcasting, DVB-T was published in 1997. With the launch of HDTV services requiring more bandwidth, an already scarce resource, a new standard that uses the latest advances in digital transmission such as channel coding and modulation was necessary. The goal of the new standard was to either increase the throughput in the given bandwidth or to use less capacity. It is for this reason that Digital Video Broadcasting Second Generation Terrestrial (DVB-T2) [5] was conceived. Compared to the first generation standard targeting HDTV, the main goals of DVB-T2 were to achieve higher bit rates, improve single frequency networks (SFNs), provide service specific robustness and target services for fixed and portable receivers [4]. It uses new modulation and error-protection to increase the efficiency of the use of the radio spectrum. DVB-T2 provides a minimum capacity increase of 30% compared to DVB-T in similar reception conditions [2]. It builds on technologies used as part of the first generation system and extends the range of most of the parameters of DVB-T and also introduces new features. These incorporated changes boost the throughput and robustness of the system, increase

coverage of SFNs and simplify the implementation of both the transmitter and the receiver [4]. DVB-T2 uses BCH and LDPC as its FEC schemes which outperform the Reed Solomon (RS) convolutional codes used in DVB-T. DVB-T2 also supports generic streams (GS) as input format.

Among other things, DVB-T2 introduces a new technique of rotated constellation which provides additional robustness to low order constellations in adverse reception conditions. The interleaving is also extended to include bit, cell, time and frequency interleaving providing more robustness. Another distinct feature introduced by DVB-T2 is the future extension frames (FEF) for compatibility with future enhancements. Furthermore, DVB-T2 introduces Alamouti coding, which is a transmitter diversity method, to improve coverage in SFNs by means of multiple antenna reception.

A new mechanism of transmitting data streams in separate logical channels is also introduced in DVB-T2. The channels are known as physical layer pipes (PLPs). PLPs allow the robustness of each service to be adjusted separately to meet the required reception conditions, also allowing power saving by allowing a receiver to decode only the desired service instead of the entire multiplex. Modulation, channel coding and interleaving can be applied separately for each PLP. This allows different reception scenarios: fixed, portable and mobile, to be targeted in the same channel. Both DVB-T and DVB-T2 uses coded OFDM modulation to overcome multipath fading and delay spread and to provide SFNs. However, OFDM suffers from high peak to average power ratio (PAPR) and unlike DVB-T; DVB-T2 provides mechanism for PAPR reduction. The peak amplifier power rating can be reduced by 25% which results in power efficiency [6]. DVB-T2 provides increased flexibility in choice of system parameters such as code rates, modulation, channel bandwidth and OFDM parameters (FFT size, guard interval duration, number of carriers and carrier mode). Table 1.1 shows a comparison between DVB-T and DVB-T2 and it can be seen that DVB-T2 has more configuration options and new features which makes it more flexible than DVB-T. The focus of this dissertation is on DVB-T2 and more details on DVB-T2 are given in chapter 2, Section 2.1.

Table 1.1 A Comparison between DVB-T and DVB-T2

Feature	DVB-T	DVB-T2 (new/improved features are in bold)
FEC	Convolutional codes + Reed Solomon 1/2, 2/3, 3/4, 5/6, 7/8	LDPC +BCH 1/2, **3/5**, 2/3, 3/4, **4/5**, 5/6
Guard Interval	1/4, 1/8, 1/16, 1/132	1/4, **19/128**, 1/8, **19/256**, 1/16, 1/132, **1/128**
Modulation Modes	QPSK, 16QAM, 64QAM	QPSK, 16QAM, 64QAM, 256QAM
FFT Size	2k, 8k	**1k**, 2k, **4k**, 8k, **16k**, **32k**
Scattered Pilots	8% of total	**1%**, **2%**, **4%**, 8% of total
Continual Pilots	2.0% of total	**0.4%-2.4%** of total (0.4%-0.8% in 8k-32k)
Bandwidth	6, 7, 8 MHz	**1.7**, **5**, 6, 7, 8, **10** MHz

1.2.2 ATSC

Advance Television System Committee (ATSC) was formed in the USA to explore advanced TV solutions. The first ATSC terrestrial standard, ATSC 1.0, was adopted by the Federal Communications Commission (FCC) in 1996. The standard is over 20 years old and uses old technologies. However, broadcasting needs new capabilities and increased capacity especially with the increasing demand for high definition TV. The ATSC has been working on a next generation standard, ATSC 3.0 [7] that will replace the old standard. ATSC 3.0 takes advantage of advances in technology such as error coding, modulation, interleaving, constellations and compression algorithms. The goal of this standard is to provide UHDTV and HDTV to improve television viewing experience; and improved and more flexible reception for both fixed and mobile reception. It also aims to provide more accessibility, personalization and interactivity.

ATSC 3.0 transport layer uses IP encapsulation for streaming and file delivery rather that the MPEG-TS encapsulation currently used by all other standards [8, 9]. This allows broadcasting to be part of the internet and enables creation of new services and business

models for broadcasters. It gives broadcasters flexibility and control over service they offer and it allows broadcasting to evolve at almost the same pace as the internet. Unlike ATSC 1.0 which uses 8-VSB, ATSC 3.0 uses OFDM as modulation waveform enabling it to support SFNs. Similar to DVB-T2; ATSC 3.0 uses LDPC with two different code lengths but provides a wider range of code rates (from 2/15 up to 13/15) and constellation orders (from QPSK up to 4096 QAM). It also introduces a concept of PLPs and allows the decoding of up to four PLPs per services contrary to DVB-T2 which allows one PLP or two PLPs when a common PLP is used. The ability to decode a maximum of 4 PLPs makes it possible to transmit layered media such as scalable videos with ease providing different error protections to each quality layer. A new frame structure is introduced which not only allows different channel coding, modulation scheme and interleaving parameters to be configured for each service, but also allows FFT sizes to be configured separately for each service. To maximise data rates ATSC 3.0 also introduces channel bonding where two RF channels are combined to achieve greater service data rates than can be achieved in one RF channel. Channel bonding is shown in Figure 1.4 where two 20 MHz channels are combined to form a 40 MHz channel.

ATSC 3.0 introduces a number of features that are not present in other terrestrial standards. Apart from channel bonding, layered division multiplexing (LDM) is introduced. LDM is a constellation superposition technology that combines two streams of different power levels with independent modulation and channel coding configurations in one RF channel [7-9]. The only drawback with ATSC 3.0 is that it is not compatible with prior standard. It can be concluded that ATSC 3.0 is currently the best terrestrial television standard; it offers the most robust, most spectrum efficient and most flexible transmission options for broadcasters and the services they provide.

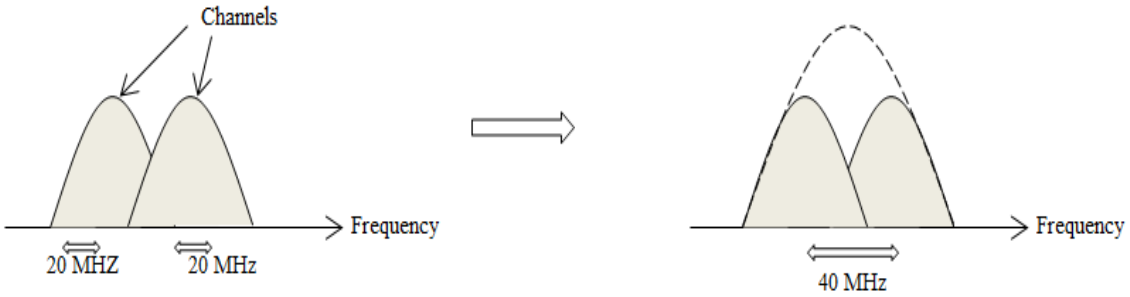


Figure 1.4 Channel Bonding Illustration

1.2.3 ISDB-T

ISDB-T was approved in 1998, long after DVB-T and ATSC1.0, and takes into account the experience gained by the standards. The system was design to provide reliable high quality video, audio and data broadcasting for fixed reception as well as mobile reception. Therefore, unlike DVB-T a low power mobile reception was built into the standard from the beginning. Like other DTTB standards, ISDB-T uses OFDM modulation particularly band segmented transmission-OFDM (BST-OFDM) which is based on COFDM. BST-OFDM allows mobile, fixed and portable reception in one RF channel. It consists of 13 OFDM segments and transmission parameters can be configured separately for each segment. An example of how a single channel is segmented into 13 segments and possible uses of the segments are show in Figure 1.5. The central segment can be used to deliver low data rate services to one segment portable receivers. Operation on a single segment results in low power consumption, making it possible to integrate a single segment receiver in a mobile phone or other handheld portable devices. There is a wide variety of configuration options for channel coding, time interleaving and modulation.

Having been conceived almost two decades ago, ISDB-T uses Reed Solomon codes and convolution codes for error correction and allows QPSK up to 64 QAM for modulation. As a result ISDB-T is not as spectrum efficient and robust as the most recent DTTB standards. By using OFDM ISDB-T allows effective use of frequency by SFNs. Simultaneous transmission of fixed-reception and mobile-reception services are made possible by hierarchical transmission. ISDB-T support hierarchical transmission of up to 3 layers (Layers A, B and C)

and transmission parameters can be changed in each of these layers allowing a wide variety of services in one channel [10].

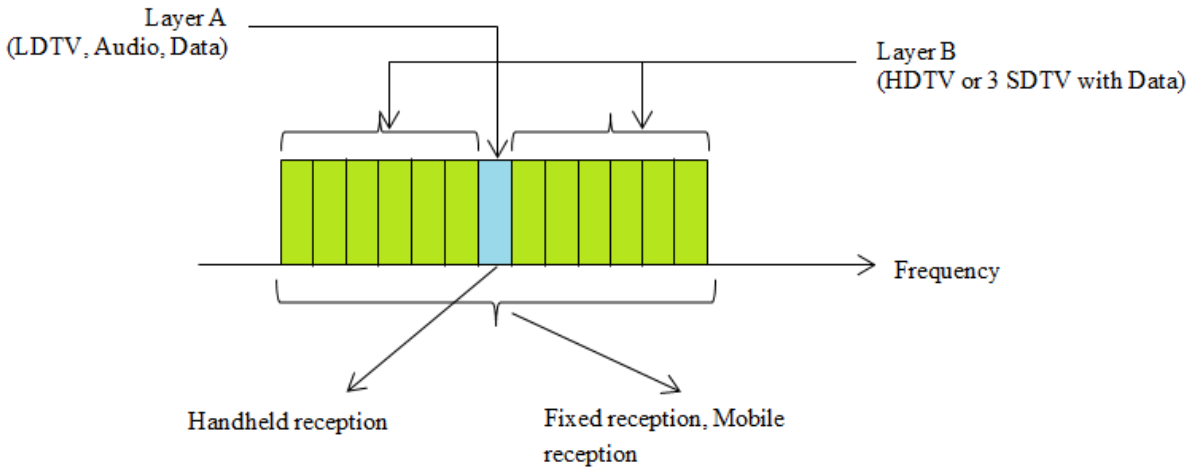


Figure 1.5 BST-OFDM Modulation Scheme

1.2.4 DTMB

The Chinese DTMB standard was released in 2006 and made mandatory in 2007. It adopts time-domain synchronous OFDM (TDS-OFDM) technology which delivers fast system synchronization, accurate channel estimation, high spectrum efficiency and excellent SFN performance. TDS-OFDM inserts a training sequence into guard intervals as the frame header and the training sequence can be used for channel estimation. DTMB adopts time-domain pseudo-noise (PN) sequence as its training sequence. Combining training sequence with guard interval not only reduces channel overhead and provides better spectrum efficiency but also good channel estimation and synchronization performances [11, 12]. The TDS-OFDM frame structure is shown in Figure 1.6. The frame header contains the PN sequence used for synchronization and the frame body which carries the actual data traffic is made up of OFDM symbols.

Being a recent standard DTMB takes advantage of recent breakthroughs in technology such as LDPC for better channel error protection, long time interleaving to reduce impulsive noise and spread spectrum protection on the system information. The standard serves both fixed and mobile receivers and supports HDTV and SDTV. In July 2015 an evolution system of DTMB named DTMB-advanced (DTMB-A) [13] was accepted by the International Telecommunications Union (ITU). This new standard increases transmission capacity by

about 30% compared to DTMB [13]. DTMB-A adopts new technologies such as near-capacity channel coding and modulation, interleaving and improved framing structure. Its supports 256 amplitude phase shift key (256 APSK), a higher modulation order than in DTMB, and this increases spectrum efficiency. It also employs improved LDPC to reduce the threshold of receiving signal SNR hence increases the robustness of the system. The new LDPC combined with APSK ensure reliability of DTMB-A in transmitting video services. DTMB-A super-frame structure introduces a channel dedicated to synchronization for fast signal acquisition, coarse timing synchronization and frequency off set estimation. This improves the systems channel estimation, synchronization and equalization performance. By adopting advance technologies DTMB-A can achieve fast channel synchronization, high spectrum efficiency and high receiver sensitivity.

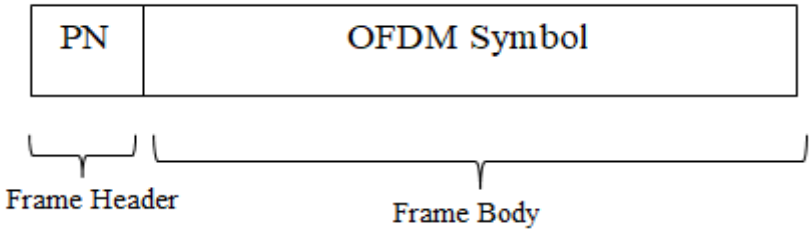


Figure 1.6 TDS-OFDM Frame Structure

1.3 Problem Statement

The emergence of high definition video such as HD and UHD and interest in 3D and 4D content comes with a need for more bandwidth. However, bandwidth is a scarce resource. Spectral efficient solutions which use less bandwidth or increase the throughput of the available bandwidth are essential. In an attempt to solve this problem, new DTT standards that use the spectrum efficiently by employing advanced technologies in modulation and channel coding were conceived and DVB-T2 is one of them. More efficient video coding techniques with reduced implementation complexity have been designed. Scalable video coding can now be achieved without a significant loss in coding efficiency and with low implementation complexity. Scalable video, unlike single layer video allows the video bit rate

to be adapted to the available bandwidth with graceful degradation in video quality. More services can be accommodated at a minimum quality and SVC also makes it possible to fill a system's capacity. Resource management procedures are also vital in improving the spectral efficiency of a system. Scheduling mechanisms, in particular, are important because they are responsible for choosing how resources are distributed in fair and efficient way between users taking into account their QoS requirements such as packet loss and delay. In this context, the design of efficient resource allocation strategies is crucial to meeting system performance targets and satisfying user requirements. An optimal trade-off between spectral efficiency and fairness has to be made while considering QoS requirements. An efficient bandwidth allocation scheme either looks to minimise the resources used or maximise the system throughput.

In DVB-T2 Scheduling is not specified; rather it is open to vendors to implement their own algorithms. Absence of definition allows more flexibility in implementation of scheduling. However, lack of standardization may result in incompatibility for hardware from different vendors and therefore hardware from different vendors may not be used in the same network because of interoperability issues. This also restricts customers to one vendor. Furthermore, without standardization the device may not work as expected yielding poor performance. For these reasons scheduling must therefore be defined for DVB-T2. Existing scheduling solutions in literature attempt to minimise the resources required to transmit the existing services in order to create room for more services. While the schemes succeed in achieving this, they overlooks the variable nature of video bit rate and limit performance evaluation to service gain and bandwidth utilization. Taking into account the variable bit rate of videos could potentially make the solution more efficient. To the best of our knowledge, allocation solutions that aim at maximising the system throughput while considering specific QoS requirements and fairness have not been investigated for DVB-T2. Allocation of bandwidth to scalable video services has also not been studied for DVB-T2. In this dissertation three PLP scheduling schemes are proposed. Two of the schemes are derived from two popular packet scheduling schemes called modified largest weighted delay first (MLWDF) and exponential proportional fairness (EXP/PF) [14-17]. M-LWDF and EXP/PF have proved to give the best performance in making a good trade-off between throughput maximization, satisfying QoS requirements and fairness particularly in long-term evolution (LTE) networks. This can be attributed to the fact that the schemes make use of more parameters when making scheduling decisions. The third scheduling scheme improves on the existing schedulers by

taking into account the bit rate variability of video. A service scheduling scheme that allocates bandwidth to SVC service by exploiting their bit rate variability is also proposed.

1.4 Research Question

The main question addressed in this dissertation is

> *How can SVC services be allocated resources in DVB-T2 in order to make a good trade-off between throughput and fairness?*

In order to address the research question the following questions are highlighted.

- How can SVC services be allocated bandwidth in order to increase throughput in DVB-T2?
- How can PLPs be scheduled in order to make a good trade-off between throughput and fairness while considering QoS requirements?
- How can PLPs be scheduled in order to meet service delay constraints satisfactorily?

1.5 Research Methodology

The following methodologies are adopted in answering the research question:

a) The focus of the research is to implement resource allocation for transmission of scalable video in DVB-T2 to improve system performance. In order to achieve this objective a thorough knowledge of DTT, DVB-T2, SVC and resource allocation, specifically bandwidth and time slice allocation, in DVB-T2 is essential. Therefore a literature review is carried out to understand the existing solutions in these areas of research.
b) System models are designed for three scheduling cases in DVB-T2:
    - Case 1: PLP Scheduling for baseband (BB) frame allocation.
    - Case 2: PLP scheduling with Service scheduling for bandwidth allocation.
    - Case 3: Service and PLP Scheduling with time slice allocation.
c) Scheduling schemes are designed and implemented for DVB-T2. Two schedulers are implemented: a service scheduler and a PLP scheduler. The service scheduler performs

input pre-processing and map scalable video to the input capacity. The PLP scheduler performs BB frame and time slice allocation.

d) The performance of the proposed schemes is evaluated against existing schemes through computer simulations in Matlab. For case 1, performance is evaluated for increasing number of services while channel over-packing is considered for cases 2 and 3. The performance is evaluated in terms of throughput, fairness and BB frame loss.

1.6 List of Publications

The following are publications associated with this research.

1. R. Khalanyane, F. Takawira and O. Olutayo, "PLP scheduling scheme for MPLP transmission of SVC services in DVB-T2", proceedings of Global Wireless Summit, pp. 175-180, Oct 2017
2. R. Khalanyane, F. Takawira and O. Olutayo, "Joint Service and PLP Scheduling with Time Slice Allocation for MPLP transmission of SVC Services in DVB-T2" to be submitted to IEEE Transactions on Broadcasting.

1.7 Organization of the Dissertation

The remainder of the dissertation is organised as follows:

Chapter 2 gives an overview of the physical layer architecture of DVB-T2 system and how bit rates and capacities for different physical layer configurations are determined in the system. The literature review on resource allocation with a focus on bandwidth and time slices is presented in this chapter. This chapter also provides an overview of scalable video coding where a description of the video coding technology is given as well as the different types of scalabilities offered by the codec. Work related to the transmission of SVC over DVB-T2 is reviewed in this chapter.

In chapter 3, PLP scheduling which focuses on the allocation of BB frame is presented. A system model, proposed PLP scheduling algorithms and simulation results are presented. Joint service and PLP scheduling is presented in chapter 4. Similar to chapter 3, a system model, service scheduling algorithms for SVC services and simulations results are provided in this chapter. The PLP scheduling performed in this chapter is the same as the one presented in chapter 3.

Chapter 5 presents joint service and PLP scheduling with time slicing for delay constrained SVC services. Like chapters 3 and 4 a system model description, time slicing algorithm and simulation results are presented. Finally, chapter 6 concludes the dissertation and introduces future work.

Chapter 2 DVB-T2 System Components

2.1 Introduction

The focus of this dissertation is on the scheduling of SVC services in DVB-T2 and this chapter provides an introduction of the two technologies. The chapter provides understanding of the DVB-T2 system and the SVC as well as literature survey on related work. In section 2.2 a detailed description of the physical layer architecture of the DVB-T2 system is provided. A literature survey on resource is also given in this section. SVC is detailed in section 2.3. The different types of scalabilities offered by the H.264/SVC are discussed and the advantages of SVC which make it suitable for DVB-T2 are outlined. A review of literature on transmission of SVC over DVB-T2 is provided in this section.

2.2 DVB-T2

2.2.1 System Architecture

The DVB-T2 system architecture is depicted in Figure 2.1. It takes as input one or more MPEG Transport Stream (TS) and/or one or more generic streams (GS). It is necessary to guarantee a constant bit rate (CBR) per service when TS encapsulation is used but the input streams are encoded with variable bit rate (VBR). To achieve a CBR, the TS flows are filled with dynamic amounts of NULL TS packets depending on the VBR encoder output. The input processing module is not part of the T2 system but may be included as a service splitter, scheduler or de-multiplexer. In [18] the input pre-processor is used as a scheduler. A scheduling algorithm for optimised mapping of service data to the data fields of BB frames to avoid fragmentation of IP packets that carry media data of high importance is proposed. The aim is to improve error resilience.

After input pre-processing the data streams are carried in logical entities called physical layer pipes (PLPs). The mapping from input streams to PLPs can be *one to one* or *many to one*. The concept of PLPs allows service specific robustness with different types of streams. Different error protection levels, including channel coding parameters, constellation order and interleaving depth can be configured for each PLP. This allows the system to be configured for example to carry PLPs with different configurations; one PLP for high data rate services (high order constellation and low protection) to broadcast HD programs for

fixed reception while the other PLP can be configured for high robustness with low data rate in order to be received by portable or mobile receivers. When multiple PLPs are transmitted the concept of common PLP can be used. The common PLP carries data shared by a group of PLPs thus eliminating the duplication of transmission of common data for every PLP. The common PLP is however not compulsory.

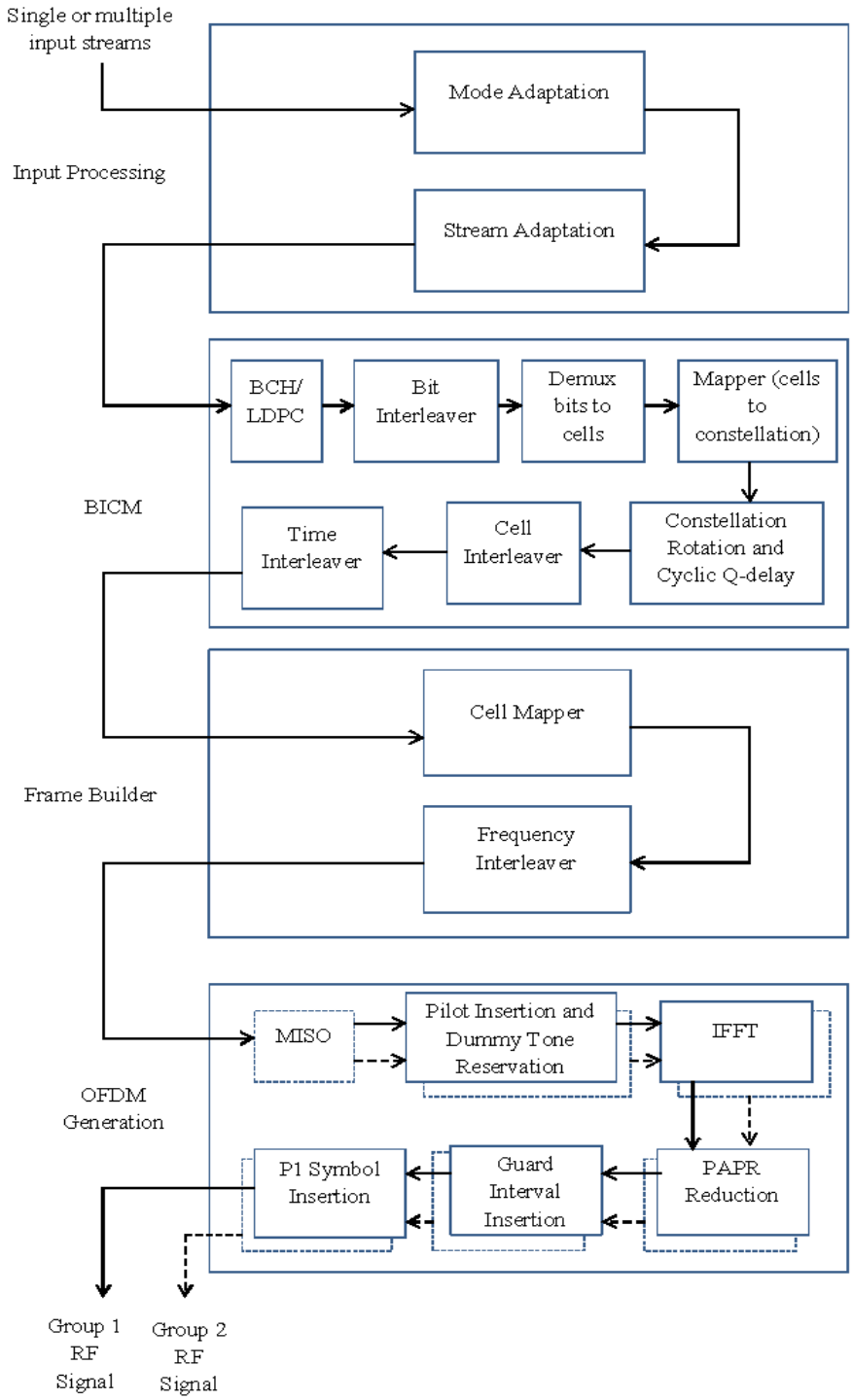


Figure 2.1 DVB-T2 system block Diagram

Following input pre-processing is the input processing module. This module is responsible for the construction of BB frames. It operates individually on each PLP. In this module PLPs go through mode adaptation and stream adaptation. The mode adaptation slices data from PLPs into data fields and a BB header is inserted at the start of each data field. The data field together with the header form a BB frame. The payload size is dictated by the FEC code rate as well as the frame length. Two frame lengths are provided; short frame which is 16200 bits long and normal/long frame which is 64800 bits long. The short frame has payload sizes varying from 7032 to 13152 bits and the long frame varying from 32208 to 53840 bits, see Table 2.1.

The mode adaptation comprises three blocks, in mode A, as depicted in Figure 2.2. The input interface generates data BB frames with maximum length K_{bch} if in-band signalling is not used. Packet fragmentation may or may not be used in the generation of BB frames. The CRC-8 block replaces the TS packet sync byte with a CRC used to detect erroneous packets. The last block, BB header insertion, adds a 10 byte (80 bits) header to each BB frame. In mode B three additional blocks are included as seen in Figure 2.3. DVB-T2 modulator may produce variable transmission delays on input data. The input stream synchroniser provides a mechanism to guarantee CBR and constant end-to-end transmission delay for any input data. When a common PLP is used, the input of a transmitter is delayed appropriately in order to allow stream recombination of data PLP and common PLP without requiring additional memory in the receiver. This is done by the compensation delay module. The null packet deletion module removes null TS packets in a data stream and replaces them by a null packet counter byte. The deleted null packets are recovered in the receiver by means of the deleted null packets (DNP) counter.

Following mode adaptation is stream adaptation (Figure 2.4). It comprises a scheduler, frame delay block, padding or in-band signalling and BB scrambling. The scheduler defines the composition of the physical layer frame by deciding which cells of the T2-frame will carry data belonging to which PLP. To do that, the scheduler generates physical layer signalling information. Padding is applied when the available data for transmission is not sufficient to complete a BB frame. The last blocks can also be filled with in-band signalling. The BB scrambler randomises each BB frame to increase information diversity.

Table 2.1 BB Frame size in bits

LDPC Code	K_{bch}	
	Short frame	Long Frame
1/2	7 032	32 208
3/5	9 552	38 688
2/3	10 632	43 040
3/4	11 712	48 408
4/5	12 423	51 648
5/6	13 152	53 840

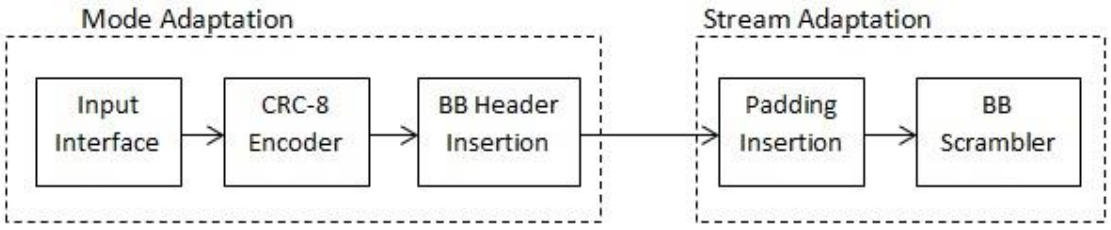


Figure 2.2 DVB-T2 Input Processing Module for Input Mode A

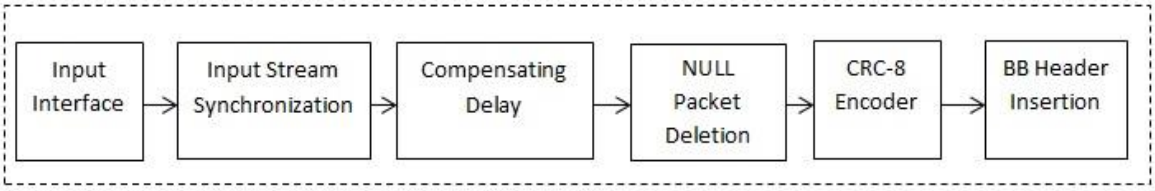


Figure 2.3 Mode Adaptation for Input Mode B

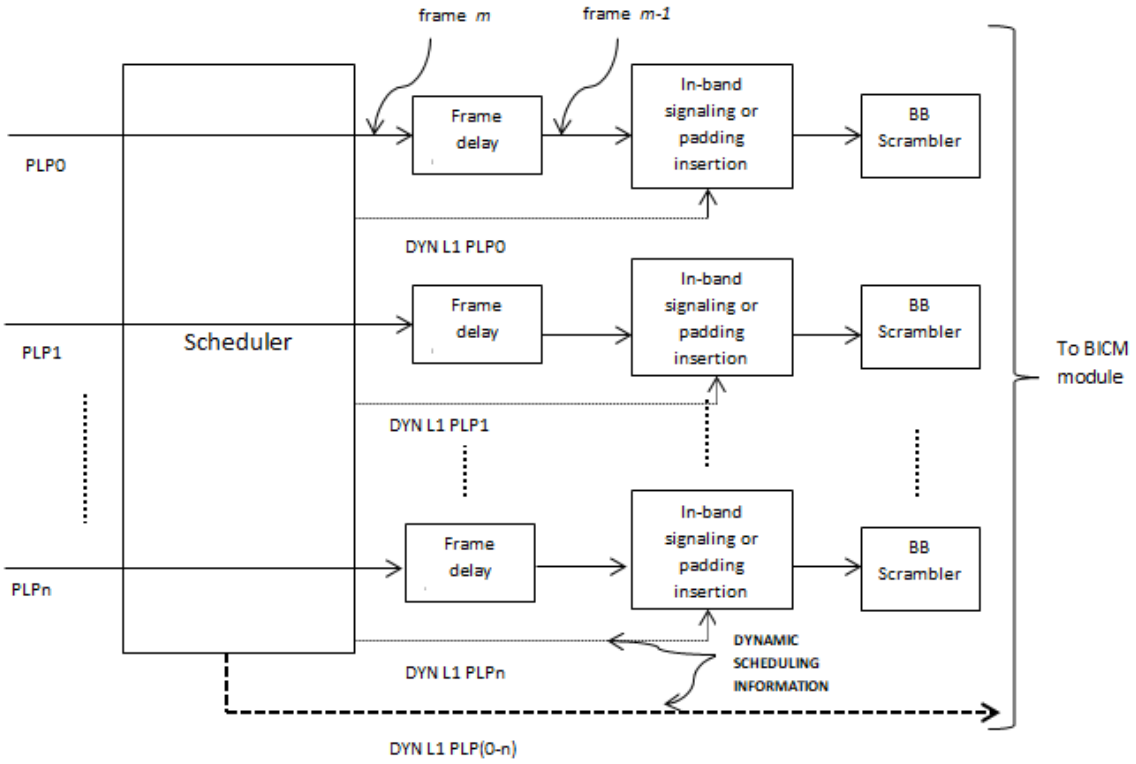


Figure 2.4 Stream Adaptation for Input Mode B

Table 2.2 Bit-Mapping into Constellation

FEC Block Length	Modulation Mode	Modulation Index	Number of Output Data Cells
64800	256	8	8100
	64	6	10800
	16	4	16200
	QPSK	2	32400
16200	256	8	2025
	64	6	2700
	16	4	4050
	QPSK	2	8100

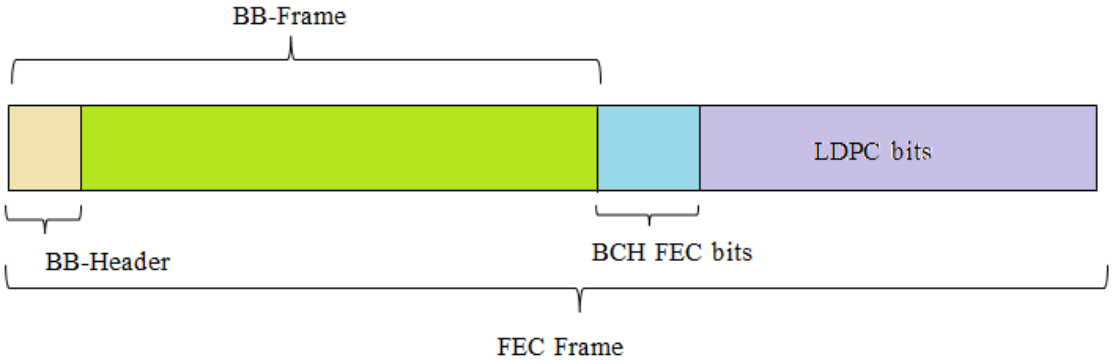


Figure 2.5 Baseband and FEC Frames Structure

After input processing, a BB frame is passed to the bit interleaved coding and modulation BICM stage. This stage is performed separately for each PLP. Chain codes are used. The outer code is BCH while the inner code is LDPC and then bit interleaving is performed on each BB frame. The input to the BCH encoder is a set of K_{bch} bits and the output comprises N_{bch} bits. The BB frame output is passed to the inner encoder and the output is refers to as a FEC frame. The BB frame and FEC structures are shown in Figure 2.5. DVB-T2 uses six code rates: 1/2, 3/5, 2/3, 3/2, 4/5 and 5/6. The code rate ¼ is used for L1 signalling. The parity bits are appended to the BB frame to create the FEC frame. The contents of FEC frames are then bit interleaved, except if QPSK is going to be used, and mapped to constellations. DVB-T2 uses QPSK, 16-QAM, 64-QAM and 256-QAM as the optional modulation schemes. The mapping of FEC frames to constellations is provided in Table 2.2. The table provides the number of output data cells FEC blocks of different lengths are mapped to for each modulation mode. This number is obtained by dividing the length of a FEC block by a modulation index associated with a particular modulation mode. Higher code rates and higher constellation orders both give high bit rates but require higher signal to noise ratios (SNR).The choice in code rate and constellation order is influenced by channel conditions.

To improve performance in frequency-selective channel imposing erasures, rotated constellations and cyclic Q-delay are employed. Figure 2.6 and 2.7 show 16 QAM constellation before and after rotation respectively. By rotating the constellation to a carefully chosen angle, each constellation can have a unique mapping onto each axis, I and Q axes, as shown in Figure 2.7. The data from each of the axes are separated in the modulator and made

to travel independently through the OFDM signal combined with I and Q data from other data cells. At the receiver, the I and Q data are recombined to give the original rotated constellation. In this way, if one of the carriers is lost due to interference, some of the signal can be recovered from the remaining axis value.

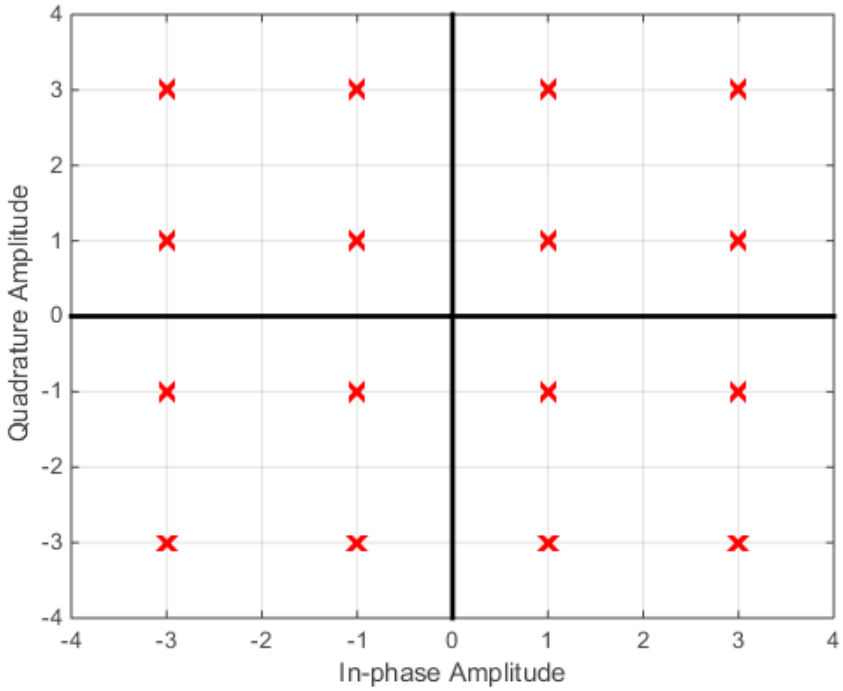


Figure 2.6 16 QAM Constellation

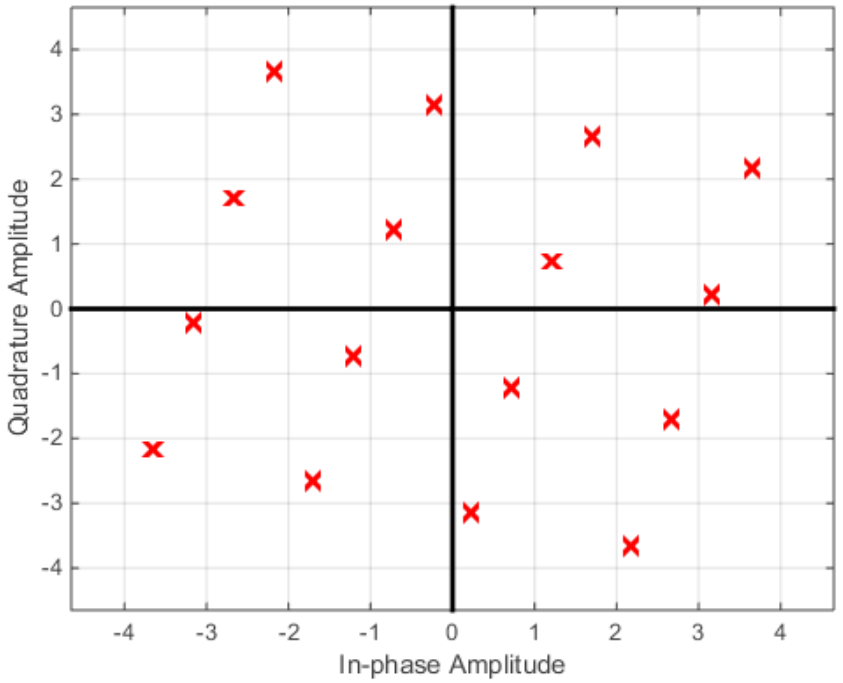
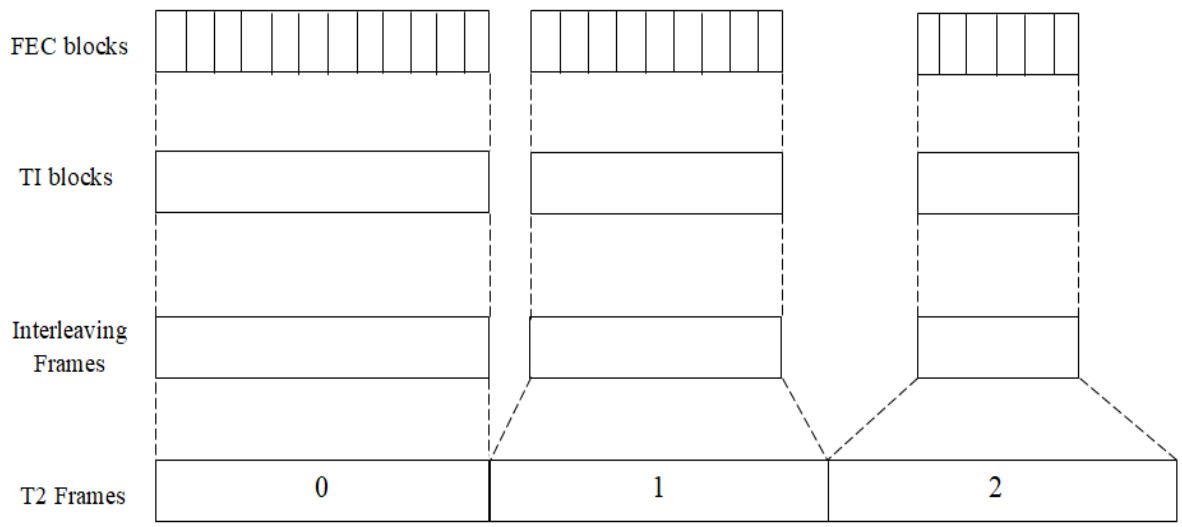
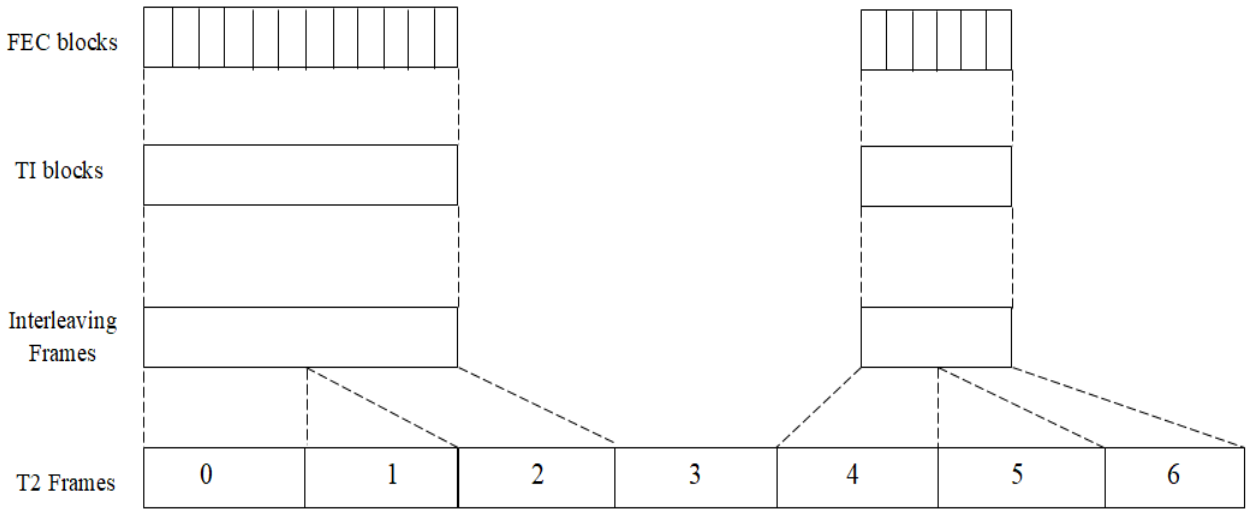


Figure 2.7 A Rotated 16 QAM Constellation

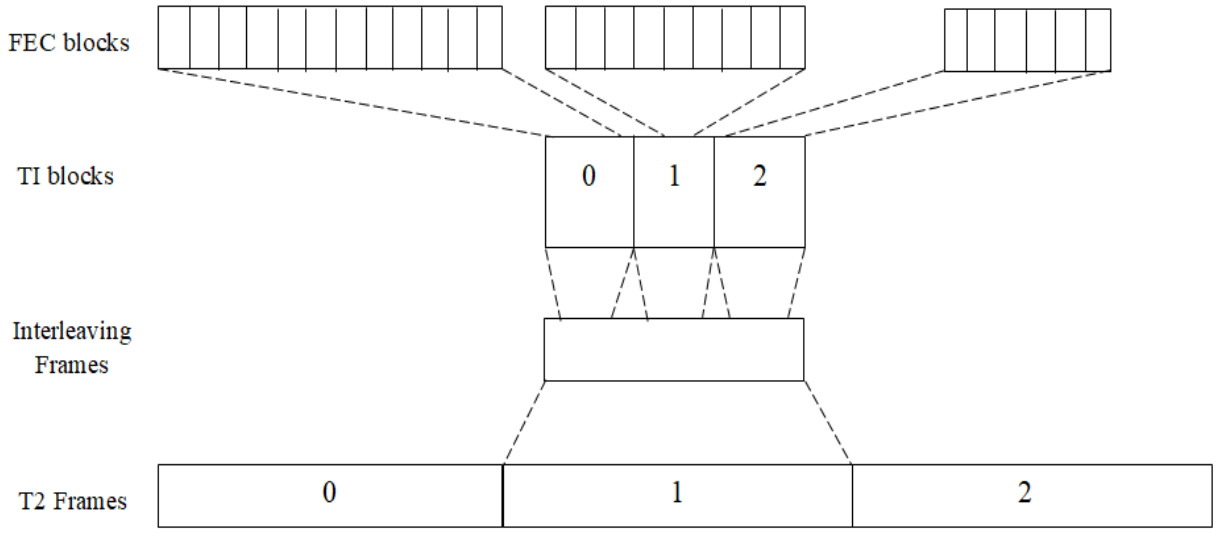
After the mapper, the symbols are cell interleaved followed by time interleaving. The cell-interleaver outputs a pseudo-random permutation of cells in a FEC frame. The aim is to uniformly spread channel errors along an entire FEC frame. The time interleaver operates on interleaving frames (IF) and time interleaving blocks (TIB). One TIB corresponds to a single usage of the time interleaving memory. The time interleaver mixes cells from different FEC frames. The standard offers three different time interleaving options depending on how FEC frames, TIBs, IFs and T2-frames are related [19]. In the first option, each IF contains one TIB and is mapped directly to one T2 frame. This is illustrated in Figure 2.8 (a). In the second option, each IF contains one TIB and is mapped to more than one T2 frame as shown in Figure 2.8 (b). The third option maps each IF directly to one T2 frame and each IF is divided into several TIBs as shown in Figure 2.8 (c). The target of time interleaving is to provide protection against impulsive noise and short time-selective fading. The time interleaving memory has a fixed size of $2^{19} + 2^{15}$ cells and acts as a buffer for PLP data prior to frame building. This dissertation focuses on scenario where each IF carries one TIB and is mapped to one T2 frame.



(a) Time Interleaving Option 1



(b) Time Interleaving Option 2



(c) Time Interleaving Option 3

Figure 2.8 Time Interleaving Options

The last modules are the frame builder and OFDM generation modules. This block is in charge of allocating the data cells from PLPs to data carriers of the OFDM symbols, the OFDM symbols in T2-frames and T2-frames in T2 super frames as depicted in Figure 2.9. The signalling information has to be allocated to this structure also. In the structure, the first

place is for preamble symbols (P1 and P2) which carry the signalling information for receiver synchronization. The start of the T2 frame is signalled by P1 symbol which is based on a 1k OFDM symbol with frequency shifted repeats at the front and rear of the symbol. The structure allows easy detection of the P1 symbol while preventing any data imitation of P1 by any part of the signal within the T2 frame. The preamble symbols are followed by data symbols. The common PLPs are mapped first, followed by data PLPs. DVB-T2 defines two types of data PLPs; type 1 and type 2. Type 1 PLPs are allocated one slice per T2 frame while Type 2 PLPs are allocated more than one slice called sub-slice in a T2 frame providing diversity in time. Type 1 frames are mapped before type 2. The arrangement of PLPs into T2 frames is described in the standard [5]. The unused data cells are occupied by dummy cells to fill the capacity of a T2 frame. The frame building is carried out according to the schedule generated by the scheduler in the input processing module.

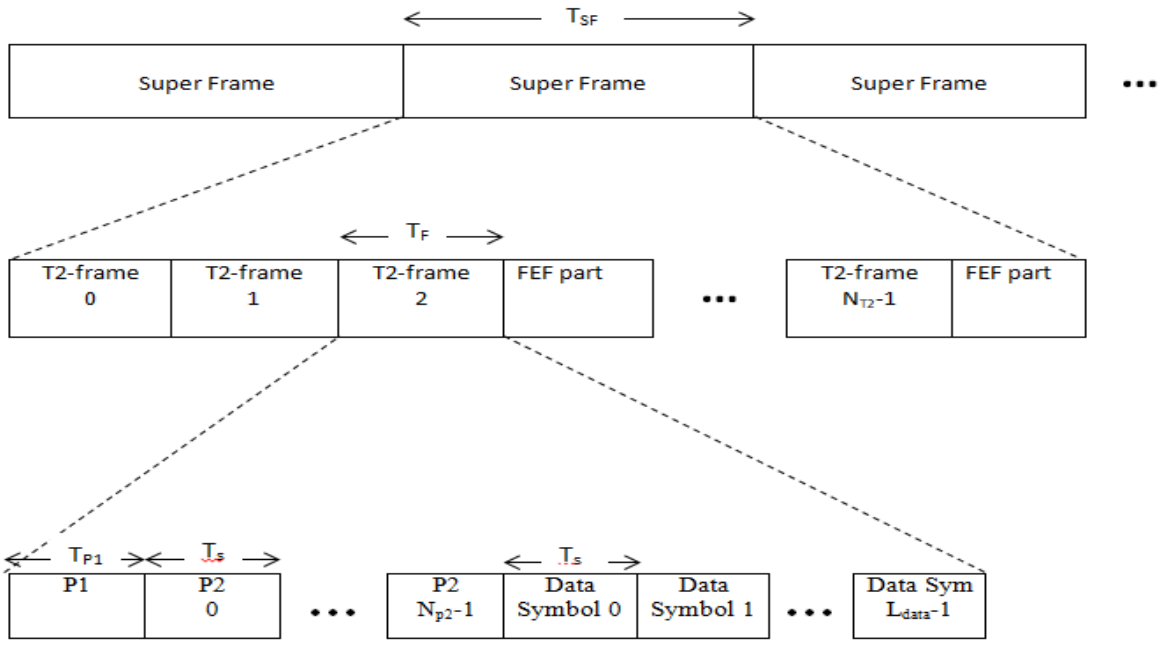
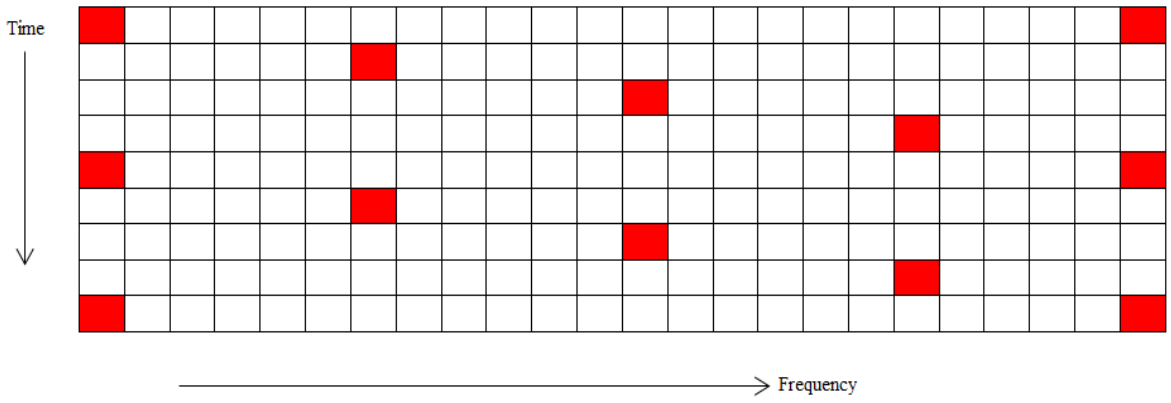
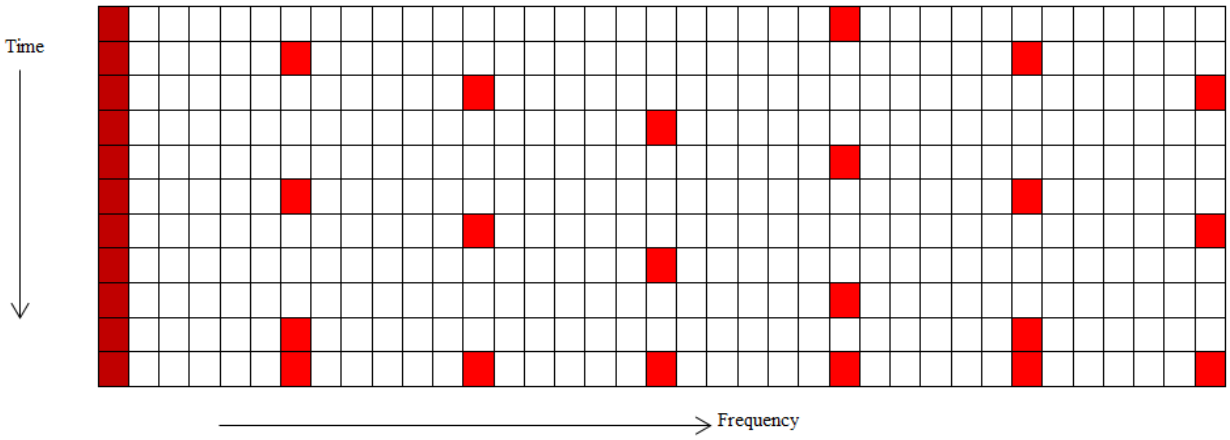


Figure 2.9 DVB-T2 Framing Structure

The output of a frame builder is followed by a frequency interleaver having a size equal to the OFDM size used. The Symbol sizes offered by DVB-T2 are 1k, 2k, 4k, 8k, 16k and 32k. The choice of FFT size is influenced by the reception scenario. Large FFT sizes have a greater vulnerability to fast fading channels hence a poor Doppler performance [6]. For fixed reception where time variations are minimised, 32k offers the highest achievable bit rate. For mobile reception, smaller FFT sizes are recommended. The frequency interleaver scrambles

the data of different PLPs as opposed to previous interleavers working on a single PLP. After frequency interleaving scattered pilots are inserted to allow synchronization and tracking, as well as channel estimation at the receiver. Scattered pilots are OFDM cells of known amplitude and phase that are used by the receiver to compensate or equalise for channel impairments as the channel changes in frequency and time. In DVB-T, scattered pilots introduce 8% overhead. DVB-T2 has 8 different scattered pilot patterns in order to minimise overhead. Figure 2.10 shows three examples of T2 scattered pilot patterns. Guard intervals are inserted and, if relevant, peak-to-average-power ratio (PAPR) reduction processing is applied to produce a complete T2 signal. An optional stage known as MISO processing allows the initial frequency domain coefficients to be processed by a modified Alamouti encoding which allows the T2 signal to be split between two groups of transceivers on the same frequency in such a way that the two groups will not interfere with each other.





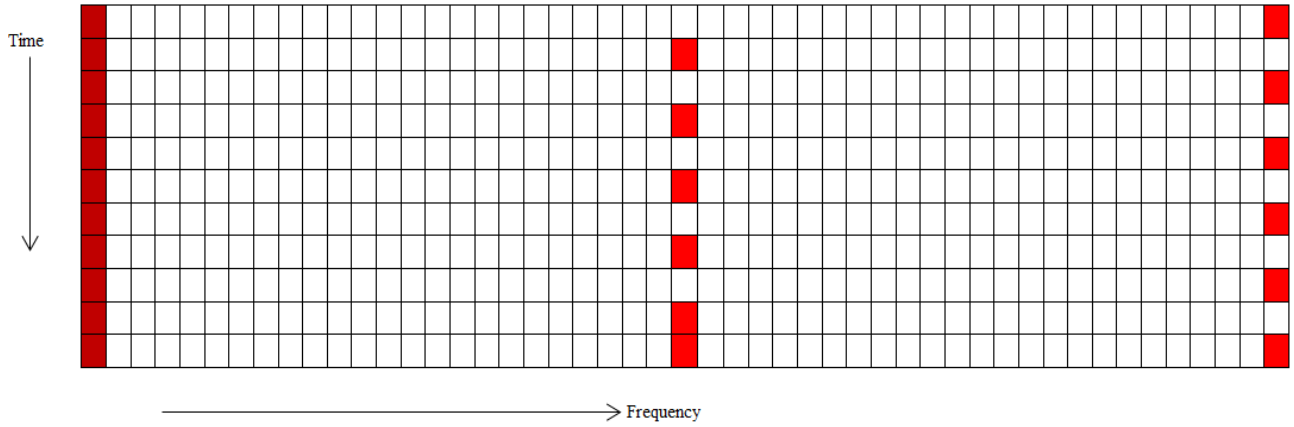


Figure 2.10 Scattered Pilot Patterns Examples

2.2.2 Capacity and Bit rates

In DVB-T2 resources are allocated in the time and frequency domain as shown in Figure 2.11. They are distributed every T2 frame. The maximum duration of a T2 frame,T_f , is 250ms. Each T2 frame is made of OFDM symbols of duration T_s. The number of symbols depends on the OFDM size in use. In the frequency domain the resources are divided into equally spaced subcarriers. The number of subcarriers also depends on the FFT size used. A modulated subcarrier is called a cell. Transmission in DVB-T2 is performed in whole number of FEC blocks therefore the minimum number of cells that can be allocated for data transmission are equivalent to the size of one FEC block in cells.

The capacity of a DVB-T2 system can be described in terms of data cell per T2 frame or the achievable bit rate per T2 frame. The following explanation closely follows the guidelines in [20]. The total capacity of T2 frame in cells C_{tot} is given by

$$C_{tot} = N_{P2} \times C_{P2} + (L_{data} - 1) \times C_{data} - C_{FC} \quad (2.1)$$

N_{P2} is the number of P2 symbols, C_{P2} is the number of cells in a P2 symbol, L_{data} is the number of data symbols, C_{data} is the number of cells in a data symbol and C_{FC} is the number of cells in a frame closing symbol. If no frame closing symbol is used $C_{FC} = C_{data}$.

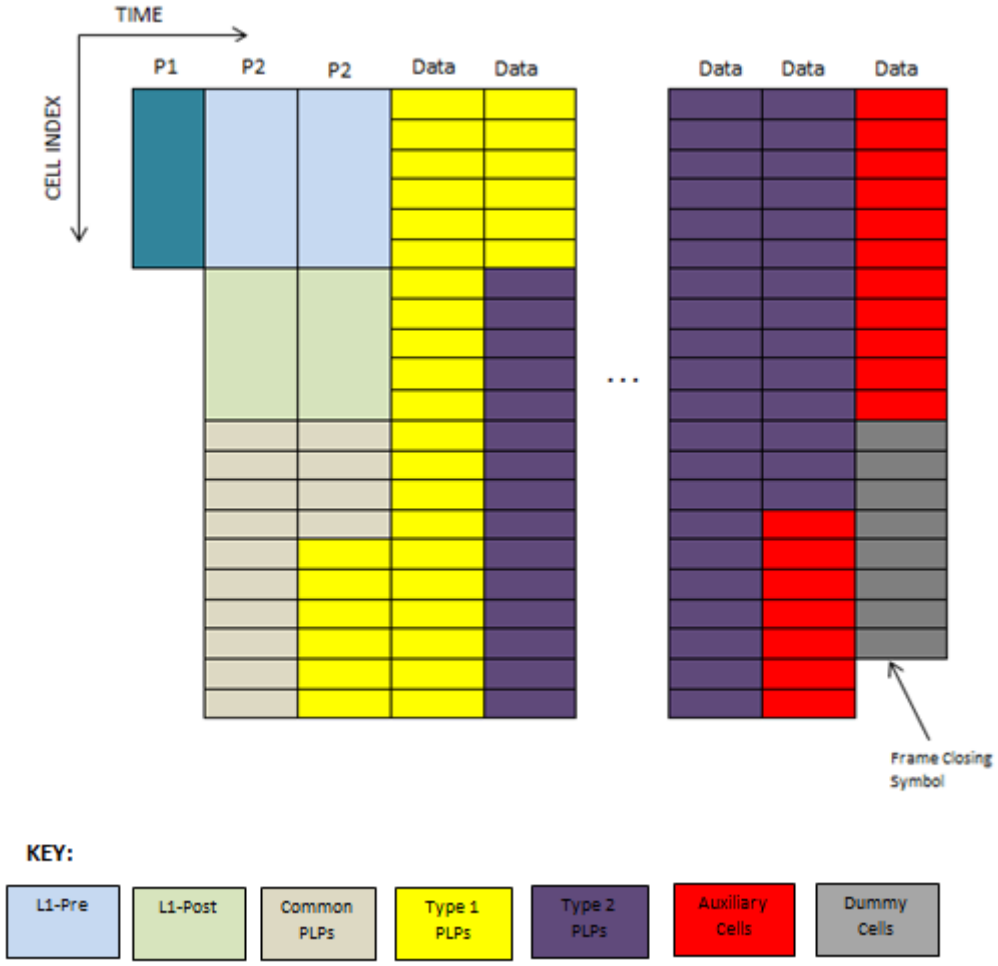


Figure 2.11 Cell Mapping in DVB-T2

To find the number of cells available for transmission of PLPs and auxiliary services, the number of cells carrying L1 signalling D_{L1} has to be subtracted per frame. Thus the number of cells available for PLPs can be calculated as

$$D_{PLP} = N_{P2} \times C_{P2} + (L_{data} - 1) + C_{data} - C_{FC} - D_{L1} \quad (2.2)$$

The number of cells is a P2 symbol, a data symbol and frame closing symbol, and if any frame closing symbol is used at all, are affected by FFT size, guard interval, pilot pattern and bandwidth.

For a given modulation scheme, the total number of FEC blocks per T2 frame N_{B_max} is given by

$$N_{B_max} = \frac{D_{PLP}}{N_{cells}} \quad (2.3)$$

where N_{cells} are the number of cells per FEC block for the given modulation order m and is given by

$$N_{cells} = \frac{B_l}{m} \quad (2.4)$$

and B_l is the length of the FEC block. Equations 2.1, 2.2, 2.3 and 2.4 are obtained from [20].

In mode B, assuming the same modulation scheme is used for all PLPs, the total number of blocks is shared between the PLPs. When different modulation schemes are used, the data cells have to be shared between the PLPs and the number of FEC blocks can be calculated for each PLP based on the modulation scheme used.

Allocation and scheduling is performed in whole number of BB frame and the data rate r_i of a given PLP in a T2 frame can be calculated using (2.5) found in [20]. The equation assumes no in-band signalling, no time interleaving, no PLP grouping and therefore no common PLPs, and mode adaptation normal mode without input stream synchronization and deletion of null packets is used. One FEC block carries exactly one BB frame adding BCH and LDPC information.

$$r_i = \frac{N_{B,i} \times \left(K_{bch,i} - 80\right)}{T_f} \quad (2.5)$$

$N_{B,i}$ is the number of BB frames allocated to PLP i, $K_{bch,i}$ is the BCH coding length which depends on the code rate and 80 is the size, in bits, of the BB frame header. The number of cell in a BB frame depends on the block length used and the constellation order configured for a given PLP. The total bit rate R of the system is the sum of the bit rates of all the PLPs in the system.

$$R = \sum_{i=1}^{M} r_i \quad (2.6)$$

where M is the number of PLPs. Equation (2.6) gives the total useful bit rate the system can carry but owing to insertion of deleted null packets, the output transport stream may have a higher rate.

2.2.3 Bandwidth Allocation

Bandwidth allocation is of paramount importance for DTT due to its scarcity yet growing demand. The challenge is that it is shared by different services with different demands that exceed the available bandwidth. The aim of bandwidth allocation is to distribute the available bandwidth to each service by splitting the available capacity among the services and guarantee of QoS requirements if the available bandwidth allows it. Bandwidth allocation goes hand in hand with scheduling. The purpose of scheduling is to distribute available resources among users in a fair and efficient way to maximise system throughput along with fairness.

Scheduling is not specified in DVB-T2; rather it is open to vendors to implement their own algorithms. Absence of definition allows more flexibility in implementation of scheduling. However, lack of standardization may result in incompatibility for hardware from different vendors and therefore hardware from different vendors may not be used in the same network because of interoperability issues. This also restricts customers to one vendor. Furthermore, without standardization the device may not work as expected yielding poor performance. For these reasons scheduling must therefore be defined for DVB-T2. Coincidentally, not much work has been done under scheduling in DVB-T2. In fact only statistical multiplexing has been investigated in [21] and [22]. Details are given in Chapter 3. In [21] the total T2 frame capacity is expressed as the total number of BB frames that can be accommodated in the frame while [22] expresses it as the total number of data cell in a T2 frame. The former assumes the same modulation order is used for all PLPs while the latter caters for different PLP configurations making it more flexible. The scheduling schemes distribute the available resources between services in proportion to their temporal bandwidth requirement. Each service is allocated an amount of resources proportional to the number of BB frames, or BB frame equivalent in cells, stored in each individual service buffer. The proposed algorithms show improved bandwidth utilization and significant gain in terms of number of services that can be transmitted compared to traditional multiplexing algorithms. However, scheduling decisions consider average service rates not varying service rates. Consequently, for parts of the service which are encoded with a higher bit rate than average, less resources than needed

may be allocated while more resources may be allocated to parts that require less bandwidth. This results in loss and wastage respectively. The performance evaluation of the statistical multiplexers is based on bandwidth utilization and service gain over deterministic multiplexing and traditional multiplexing respectively. Performance factors such as system throughput, BB frame loss and fairness are not considered.

In this dissertation a scheduling scheme which considers instantaneous service rates to account for the variability of video bit rate is proposed. Two more scheduling schemes are proposed that take into account throughput, fairness and QoS demands when making scheduling decisions.

2.2.4 Time Slice Allocation

Time slicing was first used in digital video broadcasting handheld (DVB-H) to transmit data in periodic burst with higher instantaneous bit rate relative to continuous transmission [23]. DVB-T2 adopted the concept of time slicing to transmit multiple PLPs. Variable length time slices are allocated to different PLPs. Time slicing allows battery power saving for power constrained receivers. Receivers can be active for a fraction of time while the requested slice is being received. This is only possible when one slice is allocated per frame. When power saving is not an issue multiple sub-slices provide time diversity. One of the new features of DVB-T2 is time frequency slicing (TFS). This feature is optional. TFS combines more than one RF channel into a wider virtual channel improving the overall bit rate and statistically multiplexes services over the virtual channel. This results in more improved time and frequency diversity which improve robustness over time varying channels and interference. In addition to increasing capacity TFS allows more services to be transmitted and more stable video quality. However, these benefits are achieved at the expense of receiver complexity.

In DVB-T2 resource allocation last for duration of a T2 frame and can only change at the beginning of the next frame. Long transmission intervals mean resource allocation may not meet delay constraints. The time slice allocation scheme described for DVB-T2 does not consider nor guarantee delivery delay constraints of services. Common PLPs are transmitted at the beginning of the frame following the signaling information. The common PLPs are then followed by Type 1 PLPs and Type 2 PLPs follow Type 1 PLPs. Consequently, the scheduler may allocate sub-carriers later than the delay constraint of a service. Authors in [24] propose a time slice allocation for multi-services in DVB-T2 which takes delay

constraints into account. Data cells of a T2 frame are divided into equal sized slices and the scheme utilizes Type 2 PLPs to achieve diversity. The data slices are organized in a binary tree and services are allocated tree nodes that satisfy their data rates and delay constraints. Terrestrial channels conditions vary rapidly and this poses a challenge in resource management. By dividing the data cells into slices and mapping a service to more than one slice, the scheme provides diversity against time or frequency selective fading while satisfying delay constraint. Simulation results show that the proposed solution is more efficient and reliable than the current resource allocation in DVB-T2. In this dissertation the time slice allocation proposed in [24] is implemented to satisfy delay constraints of scheduled services.

2.3 SVC and Resource Mapping

2.3.1 Scalable Video Coding

H.264/ Scalable video coding (SVC) was designed as an extension to H.264/Advance video coding (AVC) standard that enables transmission and decoding of partial bit streams while retaining reconstruction quality that is high relative to the rate of partial streams [25]. It allows three types of scalability: temporal, spatial and quality or SNR scalability. A combination of all three scalability functionalities within one bit streams is called combined scalability. A scalable bit stream consist of one base layer and one or more enhancement layers. The base layer provides basic quality while the enhancement layers improve on the quality. The base layer is fully backward compatible with the H.264/AVC bit stream. The more layers received the better the quality of experience the user has. Prior video coding standards define scalable profiles but were rarely used because scalability came with significant loss in encoding efficiency and a large increased decoder complexity [25]. H.264/SVC has an improved coding efficiency and degree of scalability as compared to prior standards. The reason for the lack of efficiency in prior standards is because of the recursive structure of the prediction loop which results in a drift problem when incomplete information is decoded [26]. The drift effect occurs when the video decoder does not have access to the same reference information used in the encoder for prediction. H.264/SVC employs advanced methods for efficient enhancement layer prediction to minimise the drift effect and improve coding efficiency. The standard has been adopted as one of the video codecs for DVB broadcast services [3].

SVC was developed as an alternative to simulcast and is more bandwidth efficient in comparison to simulcast [27, 28]. Instead of sending different version of the same content to serve multiple receivers with different capabilities as in simulcast, an SVC bit stream may be transmitted to address the needs of all those receivers. This is shown in Figure 2.12 and 2.13 respectively. With simulcasting, copies of the same video clip are encoded separately for different channel bandwidths and receiver capabilities. With SVC, one copy of the video clip is encoded once and different video layers are transmitted through channels with different bandwidths and the receivers only decode the video layers they are capable of receiving. SVC bring additional benefits including graceful degradation in a lossy environment as well as bit rate, format and power stream adaptation which can be exploited under network limitations [25].

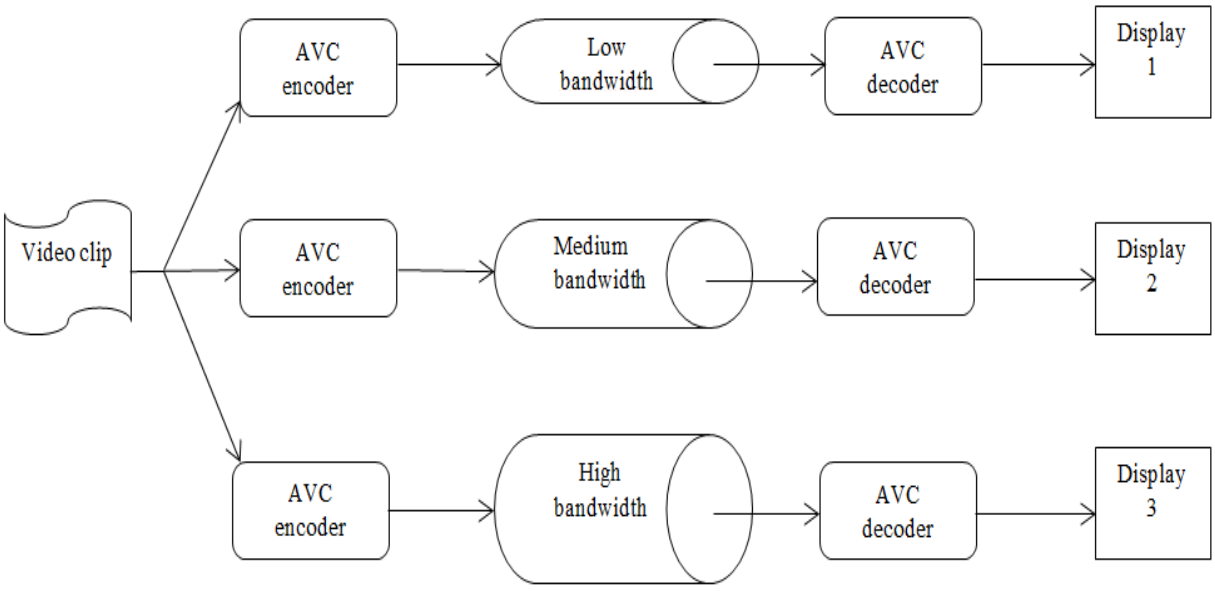


Figure 2.12 Illustration of Simulcasting

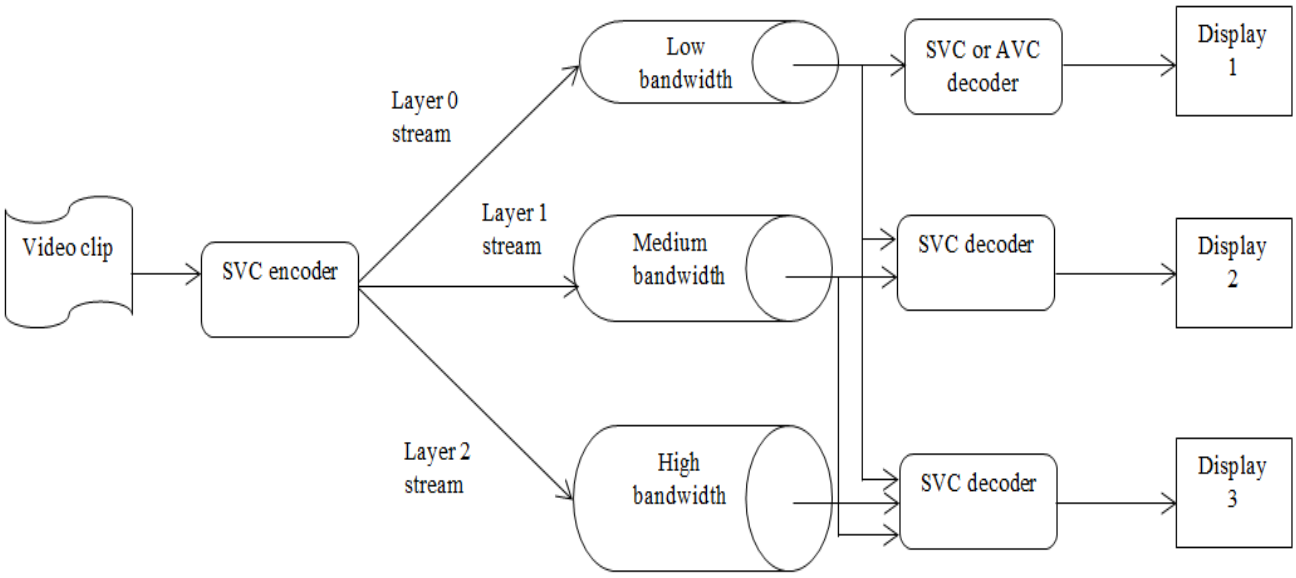


Figure 2.13 Illustration of Scalable video Coding

One benefit of SVC is that for heterogeneous receivers, the source can only be encoded once for the highest required resolution and bit rate resulting in a scalable stream from which presentations with lower quality can be obtained by discarding selected data. Thus in a multicast or broadcast scenario, terminals with different capabilities are served with a single scalable bit stream. SVC shows better bandwidth saving in comparison to simulcast and single layer video. Simulation results in [28] show that more services can be accommodated in the same bandwidth when SVC is used because of its higher coding efficiency. SVC also allows unequal protection to be applied to different layers of the bit stream providing stronger protection to the most important parts for more error resilience. For example, the base layer can be transmitted with stronger protection than enhancement layers because the base layer is essential in the decoding of enhancement layers. Loss of base layer packets results in the loss of an entire frame the packets belong to. In a bandwidth limited network, the bit rate of an SVC data stream can be adapted to match the channel throughput providing the best quality subject to network limitations.

2.3.2 Types of Scalability

Scalable video coding provides three types of scalability temporal, spatial and quality or SNR scalability. All three scalabilities can be combined forming what is called combined

scalability. That is, within one spatial layer there can be one or more quality and temporal layers.

Temporal Scalability

Temporal scalability refers to the ability to represent video content with different frame rates by as many bit stream subsets as needed. Figure 2.14 gives an illustration of temporal scalability where a video bit stream comprises of two subset bit streams with different frame rate. In the example, the base layer is encoded at 15 fps and the enhancement layer at 30 fps. Receiving only the base layer results in frame rate of 15 fps and receiving both the base layer and the enhancement layer results in video with a frame rate of 30 fps. It uses hierarchical prediction structure to improve flexibility. This means temporal scalability is not achieved by simply dropping random frames of a video sequence but encodes the frame in a certain manner. The hierarchical prediction structure improves the information between consecutive frames and allows both dyadic and non-dyadic temporal scalability.

Spatial Scalability

Spatial scalability represents through a layered structure, videos with different resolutions i.e. the enhancement layer is responsible for improving resolution of lower layers. This is shown in Figure 2.15. The base layer is encoded with a lower resolution than the enhancement layers and the enhancement layer with the highest index gets the highest resolution. Decoding only the base layer results in low picture resolution and decoding more layers results in increased picture resolution. H.264/SVC introduces a complex prediction module called inter layer prediction (ILP). The goal of this module is to increase the amount of reused data in the prediction from inferior layers so that the reduction of redundancies increases the overall efficiency. IPL reuses motion vectors, intra-texture and residue information among subsequent layers for prediction [26].

Quality Scalability

Quality/SNR Scalability transmits complementary data in different layers in order to provide videos with different quality levels as shown in Figure 2.16. A lower quality is achieved by decoding only the base layer and full quality is obtained when all layers of the video sequence are decoded. Different quantization parameters are adapted for each layer. H.264/SVC provides three types of quality scalability [26].

- **Coarse Grain Scalability (CGS):** Each layer has an independent prediction procedure. It uses the same inter layer prediction as spatial scalability but all layers

have the same resolution. CGS allows a few bit rates to be supported in a stream and the number is identical to the number of layers. It provides scalability by dropping complete layers.

- **Medium Grain Scalability (MGS):** It uses a more flexible prediction module where both the base and enhancement layers can be referenced. Drifting effect can be induced when only the base is received. To mitigate this MGS uses a concept of key pictures for resynchronization. Finer granularity level of quality scalability is provided by partitioning a given enhancement layer into several MGS. Individual layers can be dropped for quality adaptations.

- **Fine Grain Scalability (FGS):** It provides adaptation of the output bit rate to the network bandwidth. It uses a technique where different layers are responsible for transporting distinct subsets of bits corresponding to each data point. This allows data truncation at any arbitrary point.

Both MGS and FGS increase the granularity of quality scalability by allowing bit stream adaptation at network abstraction layer (NAL) unit basis. This enables packet based quality scalable coding because NAL units can be discarded from the enhancement layer of a quality scalable bit stream as compared to CGS which drops complete quality layers. NAL defines the encapsulation of the coded video data for transportation and storage.

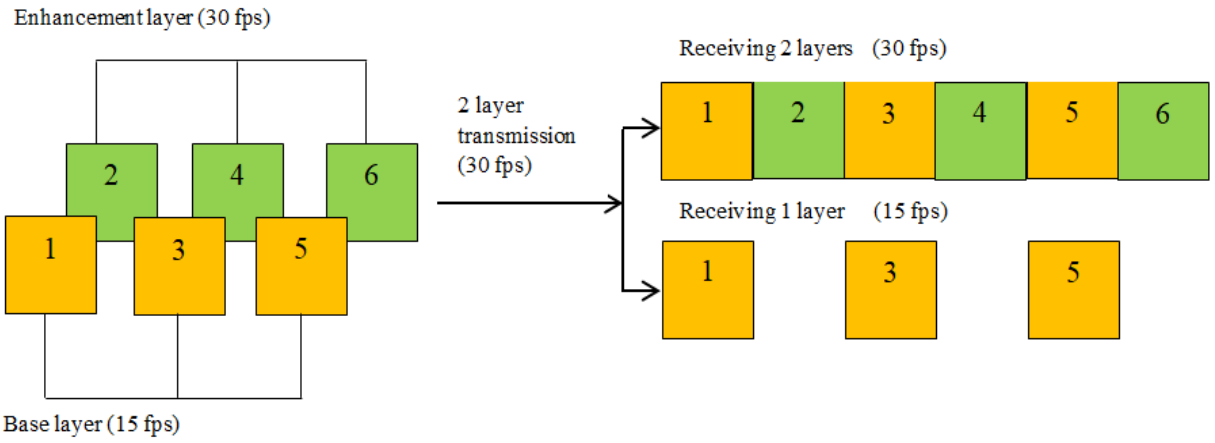


Figure 2.14 An Example of Temporal Scalability

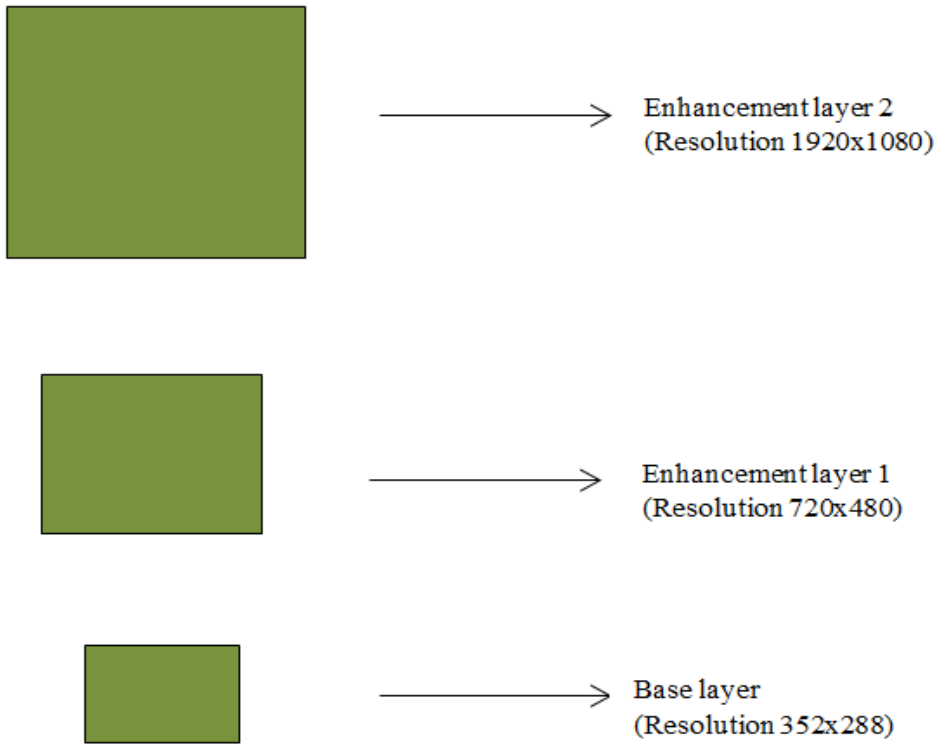


Figure 2.15 An Example of Spatial Scalability

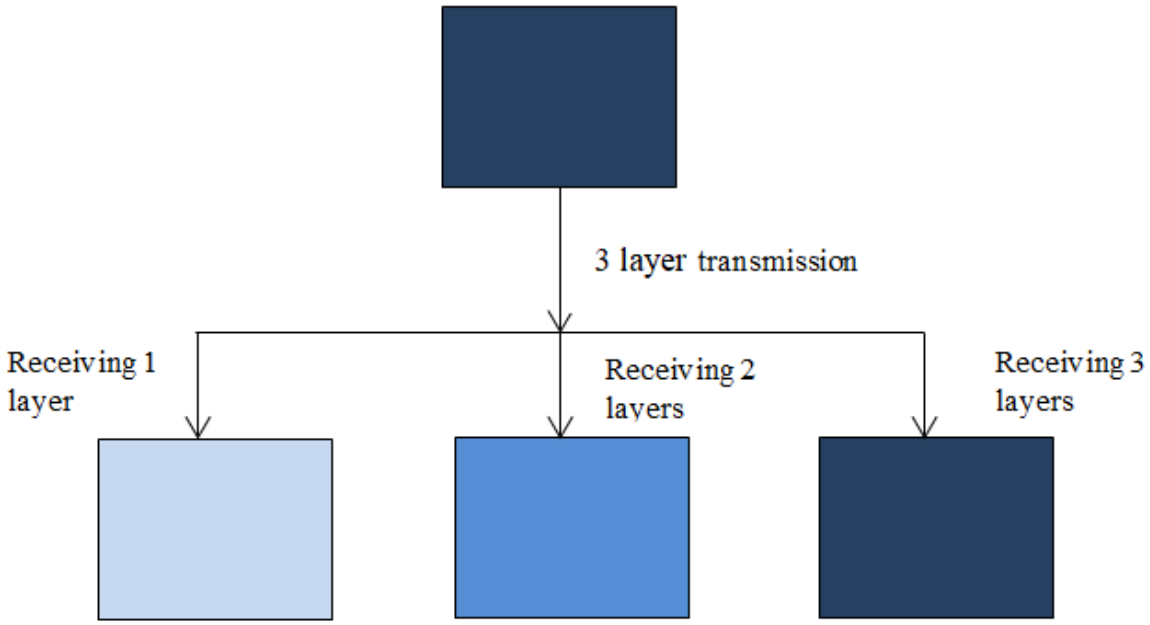


Figure 2.16 An Example of Quality Scalability

2.3.3 Resource Mapping – Related Work

Transmission of SVC over DVB-T2, for mobile and fixed reception, has been investigated in [18, 28-30]. The introduction of PLPs in DVB-T2 enables transmission of services with different protection levels and allows different reception conditions to be catered for in one multiplex. By representing a video stream in multiple layers SVC enables heterogeneous receivers to be served by one video stream. Combining the concept of PLPs and SVC in DVB-T2 can improve the efficiency of DVB-T2 in terms of bandwidth utilization and quality of experience (QoE) for users. Different protection levels can be applied to the different video layers with the base layer getting the strongest protection so that it can be decoded by even the users with the worst channel condition. This way the users with the worst channel condition can achieve basic video quality meanwhile the users with the best channel condition can decode all SVC layers and benefit from high video quality. It should be noted that legacy terminals are only capable of receiving one data PLP at a time or a data PLP and its common PLP.

In [18] three strategies have been proposed for transmitting SVC over DVB-T2. The first approach is to transmit all the layers of a scalable bit stream in one PLP. The receiver is forced to receive the entire stream and may discard the irrelevant data. This imposes a power processing penalty on power constrained terminals. The second approach schedules the video layers such that the base layer is transmitted in odd frame while the enhancement layer is transmitted in even frames. This would require the appropriate signalling information to be generated for terminals that require only the base layer to discard frames containing the enhancement layer. The drawback of the two methods is that all layers are transmitted with the same modulation and code rate thus the same error protection. A third approach transmits each stream layer in a separate PLP thus enabling different protection levels for the base layer and the enhancement layer. However, DVB-T2 specification allows reception of one data PLP and its associated common PLP, if any. The method presented would require the receiver interested in enhancement layers to be able to receive multiple data PLPs simultaneously. In [28-30] this third approach is implemented using a data PLP and a common PLP. The common PLP is used to carry the enhancement layer while the data PLP carries the base layer. This ensures compatibility with legacy terminals. Applying different protection layers to the quality layers ensures that the users with good channel condition benefit from high video quality and users with the worst channel condition achieve an acceptable video quality. Simulation results

in [28] show that less bandwidth is used when SVC is adopted as compared to simulcasting when delivering the same services. Services require less bandwidth thus more services can be transmitted in the same bandwidth. There is, however, no investigation that looks at transmission of scalable video services over DVB-T2 under capacity limitations. In this research, service scheduling for scalable video services is investigated. The service scheduling exploits the bit rate adaptability of SVC encoded videos by allocating capacity to only a subset of layers when all layers cannot be transmitted. The aim is to ensure that the highest possible video quality can be achieved in the available bandwidth.

2.4 Chapter Summary

In this chapter, an overview of the DVB-T2 system and SVC are presented. A detailed description of the physical layer architecture of the DVB-T2 system has been provided in order to understand how the system operates. Advantages of SVC that make the codec favorable over simulcasting are highlighted. In telecommunication systems, there is a need for efficient use of resources hence RM schemes, particularly scheduling, have to be put in place. From the literature review it can be seen that not a lot of work has been done in this regard and the investigations that exist have some limitations. Moreover, scheduling has not been standardized for DVB-T2 and this comes with disadvantages mentioned in Section 1.3. There is therefore a need for standardization and efficient RM schemes. This forms the basis of this dissertation.

Chapter 3 PLP scheduling

3.1 Introduction

The subject of resource management, particularly scheduling, has not been thoroughly investigated for DVB-T2. The existing bandwidth allocation schemes limit performance to service gain and bandwidth utilization. The major challenges in RM for DVB-T2 are the lack of return channel and long transmission intervals. As a result, it is difficult to make use of CSI in scheduling, and long transmission intervals and a terrestrial channel also makes it difficult to use CSI even if a return channel was present. In this chapter existing packet scheduling schemes, MLWDF and EXP/PF, used in other wireless communication systems such as long-term evolution (LTE) are adopted for DVB-T2. The two packet scheduling algorithms show good overall performance in LTE and consider many parameters when making scheduling decisions and take into account QoS requirements. An improved statistical multiplexing scheme called dynamic statistical multiplexing (D-StatMux), which takes into account the variability of the video bit rate, is proposed. The performance of the proposed schemes is evaluated against existing solutions for DVB-T2. In Section 3.2 a system model for PLP scheduling in DVB-T2 is described. Section 3.3 describes the proposed scheduling schemes. The simulation results are provided and analyzed in Section 3.4 and Section 3.5 provides the chapter summary and concludes the chapter.

3.2 System Model

3.2.1 System Model Description

The reference scenario is represented by the system block diagram as depicted in Figure 3.1 where multiple scalable video services have to be scheduled for transmission in DVB-T2.The capacity of the system depends on physical layer parameters. Each scheduled video service is mapped to its own PLP. The mapping is performed randomly. The PLPs are allocated capacity in T2 frames by scheduling to maximise both system and user throughput while maintaining fairness. The receivers of the video services are assumed to be heterogeneous with some user only capable of decoding the base layer while others can decode the base layer plus one or more enhancement layers. Because DVB-T2 is a broadcast system, each user receives all the transmitted videos, extracts the desired video PLP from the T2-frame and discards the layers it cannot decode.

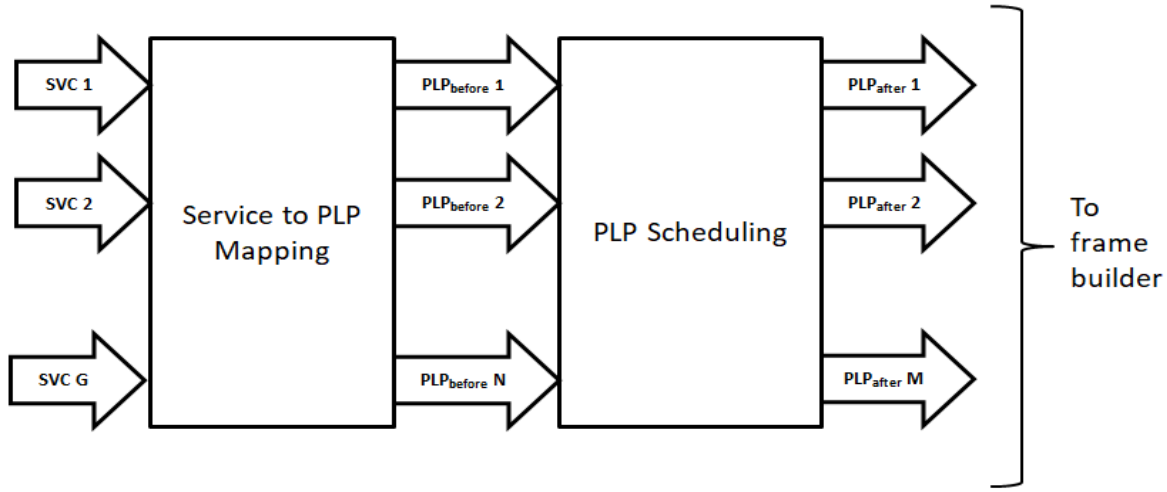


Figure 3.1 System Model for PLP Scheduling

Let G be the number of scalable videos each encoded into L layers which consist of a base layer and $L-1$ enhancement layers. $l=0$ indicates the base layer, $l=1$ is the first enhancement layer and so on. The base layer which provides basic quality and is essential for decoding the whole video frame is transmitted for every video. All layers of a video are mapped to the same PLP. Let N be the number of configurable PLPs. The set of resources units in a frame i.e. BB frames, is denoted by B_{T2}. This number will depend on OFDM parameters for the chosen reception scenario. Let the average bit rate of video g be denoted by $R_g$and $R_g(t)$ denote the instantaneous bit rate relevant to video g. Finally $b_i \subseteq B_{T2}$ represents the number of BB frames (that meet the instantaneous bit rate) selected for transmission of a PLP $i, i = 1,2, \dots, N$ carrying such a video. Not all PLPs may be allocated BB frames, therefore, let M denote the number of PLP that have been allocated BB frames. The service to PLP mapping performs a one to one mapping of G services to N PLPs randomly. PLP scheduling aims to distribute B_{T2} BB frames of a T2 frame in an efficient and fair manner between N PLPs. The scheduling is performed for every T2 frame and scheduling decisions consider buffer states, average bit rates, instantaneous bit rates, spectral efficiency, expected throughput and past throughput.

3.2.2 System Constraints

The following constraints are imposed:

a) Resources allocated in a scheduling frame shall not exceed the number of available resources.
b) A PLP shall not be allocated more resources than it needs

3.3 PLP Scheduling Algorithms

The three proposed PLP scheduling algorithms are described in this section. The dynamic statistical multiplexing algorithm, like the existing solution, perform bandwidth allocation on a T2 frame basis i.e. the algorithm is executed once per T2 frame. MLWBOF and EXP/PF on the other hand works on per BB frame basis i.e. for each BB frame a metric is computed for all PLPs and the PLP with the highest metric is allocated the BB frame. The computations of the metrics are evaluated using buffer state, resource allocation history and spectral efficiency. The service buffer conditions are used to avoid overflow by giving priority to the service which is most likely to over flow. Information from past performance is used to improve fairness by giving the highest priority to the PLP that achieved the lowest throughput in the past. For spectral efficiency, the modulation and coding values are considered to allocate resources to the services with the best modulation and coding and therefore higher expected throughput.

3.3.1 D-StatMux

Dynamic Statistical Multiplexing improves on the statistical multiplexers proposed in [21, 22]. The statistical multiplexers distribute resource between PLPs in proportion to their temporal bandwidth requirement. In [21] the T2 frame capacity for data transmission is given in terms of BB frames. Although in [21] this scheme is described as a statistical multiplexer, it is a generalized processor sharing scheme. Nonetheless we will continue referring to it as statistical multiplexing. In the scheme, time frequency slicing is considered and each TFS frame carries B_{TFS} BB frames. For N PLPs with the same service bit rate each PLP is allocated b_i BB frames of the TFS frame as

$$b_i = \frac{B_{TFS} B_i}{\sum_{j=1}^{N} B_j}, \quad i = 1,2, \dots, N. \tag{3.1}$$

Where B_i is the buffer occupancy of the service i. As BB frames are generated each PLP stores its BB frames in a finite buffer of size B_{max}. The size of the buffer is the same for all

PLPs and buffer occupancy refers to the amount of PLPs available in the buffer at an instant in time.

When the service rates are different the buffer occupancies are normalized to the average service rates as

$$b_i = \frac{B_{TFS}{}^{B_i}/_{R_i}}{\sum_{j=1}^{N} {}^{B_j}/_{R_j}}, \quad i = 1,2,\dots,N. \tag{3.2}$$

Where R_i denotes the average bit rate of PLP i and this value is the obtained during source coding and compression.

In [22] the TFS frame capacity is given in terms of amount of cells B_{cells} and the amount of information to extract from each service buffer is given as

$$b_i = \frac{B_{cells}{}^{B_i}/_{R_i}}{\sum_{j=1}^{N} {}^{B_j}/_{R_j}}, \quad i = 1,2,\dots,N. \tag{3.3}$$

The buffer occupancy is also treated in BB frames and cells respectively and the number of BB frames and cells depend on the PLP physical layer configurations. Although the algorithms are termed statistical multiplexers they are essentially generalized processor sharing (GPS) schemes.

The bandwidth allocation algorithm proposed in this dissertation assumes the instantaneous services rates for every T2 frame are known for each service. By considering instantaneous service rates, the algorithm accounts for the bit rate variability of the video services. This results in a more efficient statistical multiplexing as opposed to existing solutions. The instantaneous service rates are used to compute the bandwidth requirement for each PLP at each time interval by solving for $N_{B,i}$ given as

$$N_{B,i}\left(R_i(t)\right) = \frac{T_f \times R_i(t)}{(K_{Bch} - 80)} \tag{3.4}$$

In (3.4), $R_i(t)$ is the instantaneous bit rate for PLP i and $N_{B,i}\left(R(t)\right)$ is the number of BB frames required to achieve that instantaneous bit rate. $R_i(t)$ is measured according to the bit rate measurement algorithm defined in [31].The bit rate is averaged over a fixed time gate or

window. The gating function is moved by discrete time intervals to produce the bit rate value for each slice. The items counted can be bits, bytes or transport stream packets. The algorithm is defined as follows:

$$R(t) = \frac{S_{element}}{T} \times \sum_{n=0}^{n=N_{ts}\theta-1} N_{elements_{t-n\theta}} \quad (3.5)$$

Where N_{ts} is the number of time slices during the time gate, $T = N_{ts}\theta$ is the duration of the time gate in seconds, θ is the width of the time slice in seconds, $element$ is the fundamental unit which is being counted by the algorithm, $S_{element}$ is the size (measured in appropriate units) of the element being measured and $N_{elements}$ is the integer number of element starts which have occurred in the time slice. In this dissertation video trace files are used to evaluate the performance of the proposed resource allocation schemes and the instantaneous bit rate of the PLPs is computed from the data in the trace files according to equation 3.5. The computation of $R_i(t)$ is shown later in the chapter.

According to the system model, N services are multiplexed and each T2 frame carries B_{T2} number of BB frames (or cells depending on the level of analysis). The algorithm allocates $b_i(t)$ number of resources to the ith service as follows

$$b_i(t) = \left\lfloor \frac{B_{T2} N_{B,i}\left(R_i(t)\right)}{\sum_{j=1}^{N} N_{B,j}\left(R_j(t)\right)} \right\rfloor, \quad i = 1,2,\ldots,N. \quad (3.6)$$

where $\lfloor \bullet \rfloor$ denotes the integer floor operator.

This way, the PLP with the highest bandwidth requirement will get more resources without starving other PLPs as all PLPs get a share of the available BB frames. The amount of BB frames allocated to each PLP will vary from frame to frame.

StatMux and D-StatMux share resources proportionally between PLPs and the number of BB frame to be allocated to each PLP is computed once for each T2 frame. Hence the complexity to allocate BB frames to PLPs is $O(N)$. This means the schedulers have a linear complexity to the number of PLPs.

The next scheduling schemes are adopted from MLWDF and EXP/PF algorithms which consider factors such as packet delay, spectral efficiency, past throughput, instantaneous

throughput and several other factors when making scheduling decisions. They support guarantees of delay and minimum throughput for services with different delay bounds. Good throughput and an acceptable level of fairness are guaranteed by incorporating proportional fair (PF) rule in the algorithm. PF maximizes throughput by assigning a resource unit to the user that achieves the maximum throughput on that resource unit. This alone would result in an unfair allocation since users with poor channel condition will only get a low percentage of the resource. PF therefore uses the idea of past average throughput that acts a weighing factor of the expected data rate to ensure that users in bad conditions will be served within a certain amount of time. Apart from fairness and throughput, QoS requirements are crucial and the two algorithms provide QoS by considering packet delay. The algorithms compute priorities for users and the user with the highest priority is placed in front of the queue to get service.

3.3.2 MLWBOF

Unlike the statistical multiplexers that share resources proportionally between PLPs MLWBOF computes a metrics for all the PLPs, compares the metrics and allocates a BB frame to a PLP that achieves the highest metric on that BB frames. The metrics are computed and compared one BB frame at a time and continue until all BB have been traversed. The proposed MLWBOF is based on the MLWDF algorithm which was designed to support multimedia real time data users with varying QoS requirements within CDMA-HDR systems [17]. MLWDF determines the priority of each user k, in each interval, according to

$$k = max_i \left(\alpha_i D_{HOL,i} \frac{R_i(t)}{\bar{R}_i(t)}\right) \quad (3.7)$$

$$\alpha_i = \frac{-log(\sigma_i)}{\tau} \quad (3.8)$$

and

$$\bar{R}_i(t) = \left(1 - \frac{1}{t_c}\right)\bar{R}_i(t-1) + \frac{1}{t_c} r_i(t) \quad (3.9)$$

Where $D_{HOL,i}$ is the head of line (HOL) delay for user i at time t. HOL is the packet that resided longest in the buffer at the base station. τ is the delay threshold or deadline of user i's packets and this varies according to the type of traffic. σ_i is the maximum probability that the HOL packet delay exceeds the deadline. The user with the strongest requirements in terms of acceptable loss and deadline expiration is preferred. $R_i(t)$ is the expected data rate of user i at

time t, $\bar{R}_i(t)$ is the past average data rate of user i until time t and $r_i(t)$ is the data rate achieved by user i at time t. t_c is a time window over which fairness is imposed. MLWDF considers HOL delay and PF properties when making scheduling decisions. It assigns the highest priority to the user with a higher HOL delay and better channel condition relative to its average level. MLWBOF has been adopted from MLWDF as follows:

- In other wireless communication systems, MLWDF prefers users with the earliest deadline and lowest acceptable loss threshold to minimise packet loss due to missed deadlines. MLWBOF is designed for DVB-T2 to prioritise buffer states and acceptable loss threshold to reduce BB frame loss due to buffer overflow.
- In LTE, the data rate achieved on a resource block depends on MCS, which is determined by the CSI of a user, and preference is given to the user that can achieve the highest data rate on that resource block to increase spectral efficiency. Spectral efficiency is. In this dissertation, MLWDF for DVB-T2 has been modified to give preference to a PLP with the highest modulation index and code rate, which gives the highest data rate, for efficient use of the spectrum.
- Both MLWDF and MLWBOF maintain fairness throughput history and give the highest priority to the user or PLP with the lowest past average throughput.

MLWBOF selects a service using the following equations,

$$k = max_i \quad \left(\propto_i \frac{1}{\overline{R_i^{av}(t)}} B_i m_i c_i \right) \tag{3.10}$$

Where k is the PLP selected, m_i is the modulation index, c_i is the code rate used for PLP i, B_i is the buffer occupancy of PLP i, $\overline{R_i^{av}(t)}$ is the average throughput of PLP i over previous allocations and is defined as

$$\overline{R_i^{av}(t)} = \frac{\overline{R_i(t)}}{N_{T2}}, \tag{3.11}$$

where

$$\overline{R_i(t)} = 0.8\overline{R_i(t - T_f)} + 0.2r_i(t). \tag{3.12}$$

m_i and c_i depend on the physical layer configurations of the PLPs. In this research, a fixed modulation index and code rate is assumed for all transmissions. In (3.12), $\overline{R_i(t - T_f)}$ is the

past throughput of PLP i and depends on past BB frame allocations, $r_i(t)$ is the throughput of PLP i on the current T2 frame and N_{T2} is the index of the current T2 frame. $\propto_i$ is given by

$$\propto_i = -\frac{log\ \delta_i}{B_{max}}, \quad (3.13)$$

where δ_i is the allowable packet loss due to buffer overflow for PLP i and B_{max} is the Buffer size.

MLWBOF, on the other hand, computes a metric for each PLP on every BB frame and allocate a BB frame to a PLP that achieves the highest metric. The algorithm performs a linear search on the PLP to find the PLP that has the highest metric. The complexity to allocate the first BB frame is $O(NB_{T2})$, the complexity to allocate to the second BB frame is $O\big(N(B_{T2}-1)\big)$, and so on. Consequently, the total complexity of the algorithms is

$$O(NB_{T2} + (B_{T2}-1)N + \cdots + 2N + N) = O\left(\frac{B_{T2}(B_{T2}-1)}{2}N\right)$$

$$\approx O\big({B_{T2}}^2 N\big) \quad (3.14)$$

The algorithms have a linear complexity in the number of PLPs and a quadratic complexity in the number of BB frames.

3.3.3 EXP/PF

Similar to MLWBOF, Exponential proportional fair (EXP/PF) computes metric on each BB frame for all PLPs and allocates a BB frame to a PLP that achieves the highest metric. It was first developed to support real time services with different QoS requirements in adaptive modulation and coding and time division multiplexing [17]. EXP/PF combines the EXP rule and PF rule for decision making to guarantee the delay of RT services and maximize system throughput. It computes the priority of a user as follows.

$$k = max_i \quad \left(exp\left(\frac{\alpha_i D_{HOL,i} - \overline{\alpha D_{HOL}}}{1+\sqrt{\overline{\alpha D_{HOL}}}}\right)\frac{R_i(t)}{\bar{R}_i(t)}\right) \quad (3.15)$$

$$\overline{\alpha D_{HOL}} = \frac{1}{N}\sum_{i=1}^{N} \alpha_i D_{HOL,i} \quad (3.16)$$

N is the number of users and all other parameters represent the same terms stated in MLWDF. When the HOL packet delays do not differ much, the algorithm performs as PF.

EXP/PF algorithm has been re-derived for DVB-T2 to allocate BB frames to PLPs and computes the metric for each service using equation (3.17) given as

$$k = max_i \quad \left(exp\left(\frac{\propto_i B_i(t) \quad - \quad \overline{\propto B(t)}}{1 + \sqrt{\overline{\propto B(t)}}} \right) \frac{1}{\overline{R_i^{av}(t)}} m_i c_i \right), \tag{3.17}$$

where

$$\overline{\propto B(t)} = \frac{1}{N} \sum_{i=1}^{N} \propto_i B_i(t). \tag{3.18}$$

When buffer occupancies do not differ much, the algorithm performs as PF. EXP/PF has been adopted for DVB-T2 as follows:

- In LTE, preference is given to packets with the strongest deadline expiration. In DVB-T2, the algorithm is designed to assign the highest priority to a PLP with the highest buffer occupancy i.e. a PLP whose buffer is most likely to overflow.
- Similar to MLWBOF and MLWDF, spectral efficiency is achieved by assigning the highest priority to PLP and user that achieves the highest throughput respectively.
- For both algorithms fairness is achieved through throughput history.

Similar to MLWBOF, EXP/PF computes a metric for each PLP on every BB frame and allocates a BB frame to a PLP that achieves the highest metric. The algorithm performs a linear search on the PLP to find the PLP that has the highest metric. The algorithm has a linear complexity in the number of PLPs and a quadratic complexity in the number of BB frames. The complexity is given in Equation (3.14) as $O\left({B_{T2}}^2 N\right)$. The complexity analysis shows that the statistical multiplexers are faster than MLWBOF and EXP/PF. All the algorithms can however be easily implemented in real-time.

MLWBOF and EXP/PF incorporate buffer states together with PF qualities when determining PLP priorities. They prioritise PLPs with the highest chances of buffer overflow and better spectral efficiency relative to its average past throughput. PF makes a trade-off between fairness and spectral efficiency. Incorporating buffer state and δ_i ensures the

probability of loss does not exceed the threshold. This allows support for multiple flows with different QoS requirements.

3.4 Simulations

3.4.1 Simulation Model

Table 3.1 DVB-T2 System Parameters

System Parameters	
Bandwidth	8 MHz
FFT Size	32K
Guard Interval	1/128
Pilot Pattern	PP7
Modulation	256-QAM
Code Rate	3/5
Block Length	64800
Service Buffer Size	$2^{19} + 2^{15}$ cells

To evaluate the performance of the scheduling algorithms the DVB-T2 settings in Table 3.1 are used. A bandwidth of 8MHz and an FFT size of 32K are assumed. 265-QAM modulation scheme and a code rate of 3/5 are chosen for the DVB-T2 parameters. For the given DVB-T2 parameters the maximum number of FEC blocks per T2 frame is 229 and the maximum channel capacity is 36.156 Mbps. The simulations are performed for 50 T2 super frames with 200 T2 frames per super frame. The duration of a T2 frame is 245.84 milliseconds.

The input services are H.264/SVC video traces [32, 33] obtained from [34]. The traces characterise an encoded video by providing time stamps, frame types, frame size and PSNR for each encoded layer and each encoded frame. Trace files are for video traffic with a resolution of 352×288 , frame rate of 30 frames per second and 9984 frames. The chosen video traces represent three genres of video: sports, news and movies. The video frames are fragmented to form BB frames of payload size 38608 bits.

The traces are structured such that for each H.264 video encoding, the frame sizes in a layer trace corresponds to the sizes (in bytes) of the corresponding layer while the quality characterization in a given layer trace corresponds to the quality of the aggregate of all layers up to the considered layer. Therefore, for an encoding with a base layer and W layers there are $W-1$ quality layers indexed $w = 0,1,\dots,W$. Trace $w = 0$ gives the size X_p^0 (in bytes) and qualities Q_p^0 for each frame. Layer traces $w = 1,2,\dots,W$ give frame sizes X_p^w in layer w and video qualities Q_p^w for the aggregate of the base layer plus enhancement layers up to and including the considered layer w. The total aggregate size of encoded frames for a given frame period is obtained as

$$X_p = \sum_{w=0}^{W} X_p^w \qquad (3.19)$$

The duration of a video frame is calculated as

$$F_P = \frac{1}{F_R} \qquad (3.20)$$

where F_P is the frame period and F_R is the frame rate of the video trace. The instantaneous bit rate over a T2 frame duration, T_f, can be calculated according to equation 3.5 as follows

$$R_i(t) = \frac{1}{T_f}\sum_{m=1}^{M} X_m, \quad m = 1,2,\dots,M. \qquad (3.21)$$

$$M = \left\lceil \frac{T_\mathrm{f}}{F_P} \right\rceil \qquad (3.22)$$

and M is the number of frames per T2 frame.

Table 3.2 provides the bit rates and delay constraints of the video traces while Tables 3.2, 3.4 and 3.5 provide the bit rates of individual layers in an SVC video trace. Eight video trace files were created from each video trace file resulting in a total of twenty four video traces. The trace files were created by scrambling the main trace files to attain traces with different stochastic properties. Figures 3.2, 3.3 and 3.4 show a sample of the frame distribution of the trace files for each video category respectively.

Table 3.2 Input Services Bit Rates and Transmission Delays

Video Trace	Frame Rate (fps)	Number of Layers	Average Rate (Mbps)	Peak Rate (Mbps)	Transmission Delay (ms)
Tokyo Olympics	30	3	1.65	12.36	175
BBC News	30	4	3.44	16.06	245
Terminator	30	4	1.85	16.15	105

Table 3.3 Bit Rates of Tokyo Olympics Video Layers

Layer	Average Rate (Mbps)	Peak Rates (Mbps)
0	0.134	2.480
1	0.332	4.120
2	1.184	5.760

Table 3.4 Bit Rates of BBC News Video Layers

Layer	Average Rate (Mbps)	Peak Rates (Mbps)
0	0.061	0.796
1	0.161	1.814
2	0.448	3.820
3	2.426	9.630

Table 3.5 Bit Rates of Terminator Video Layers

Layer	Average Rate (Mbps)	Peak Rates (Mbps)
0	0.042	0.299
1	0.502	1.421
2	0.350	4.000
3	0.956	10.430

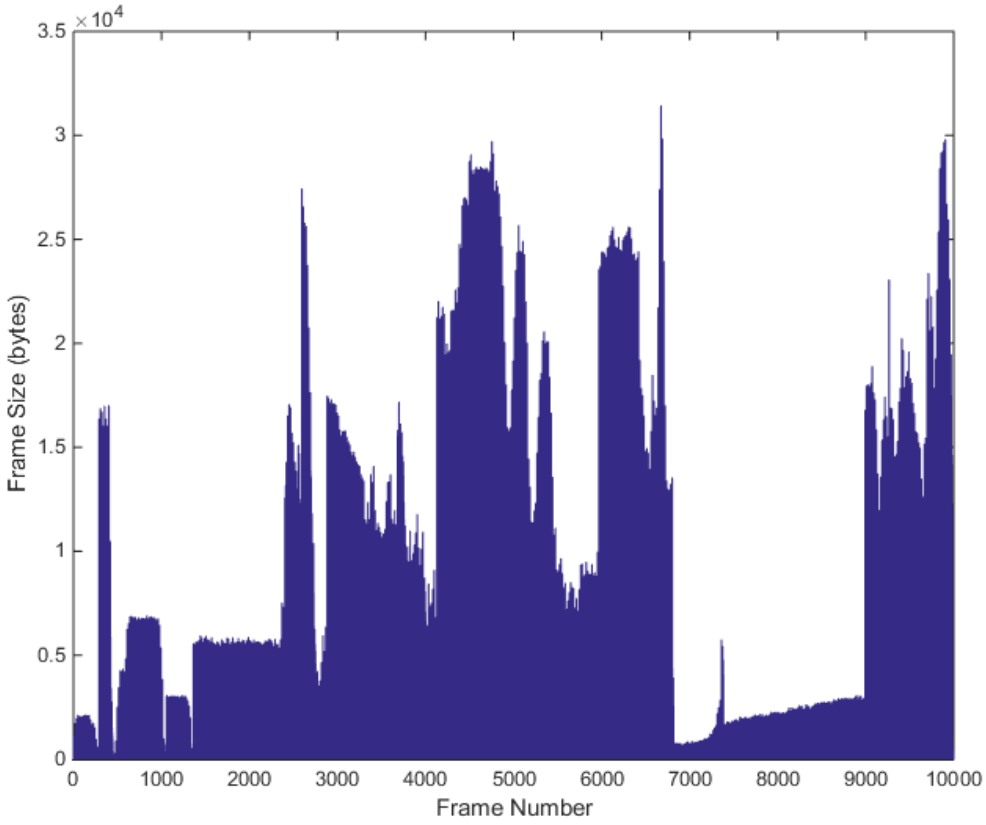


Figure 3.2 Frame Distribution Graphs for Sport Trace Files

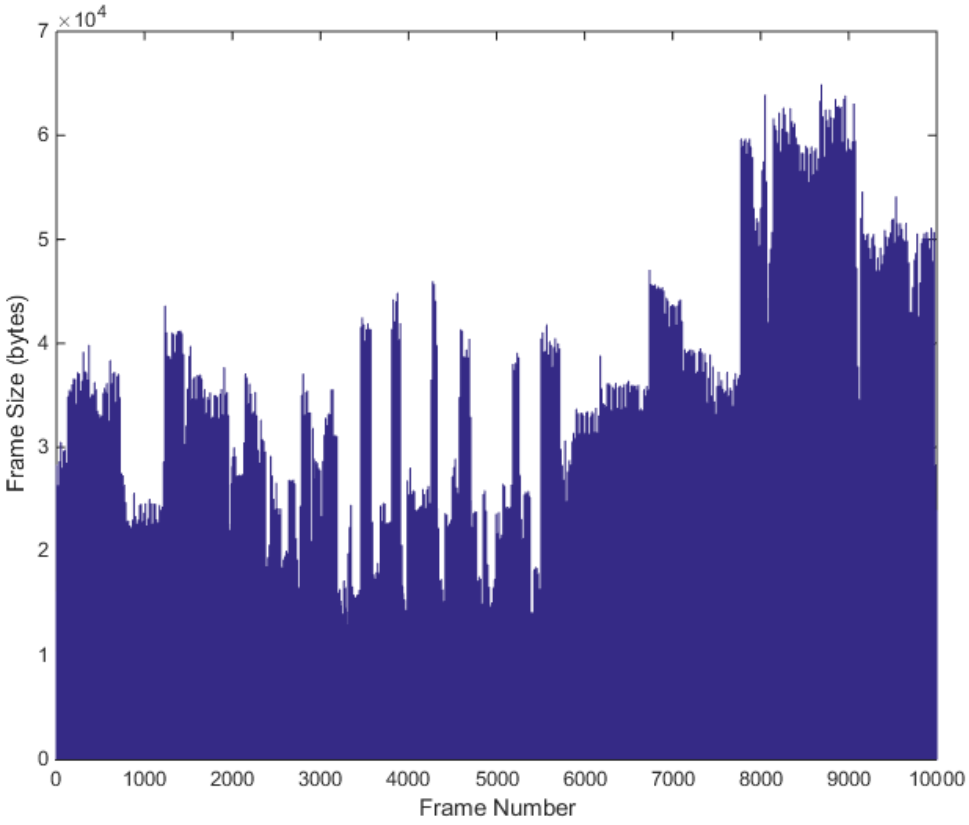


Figure 3.3 Frame Distribution Graphs for News Trace Files

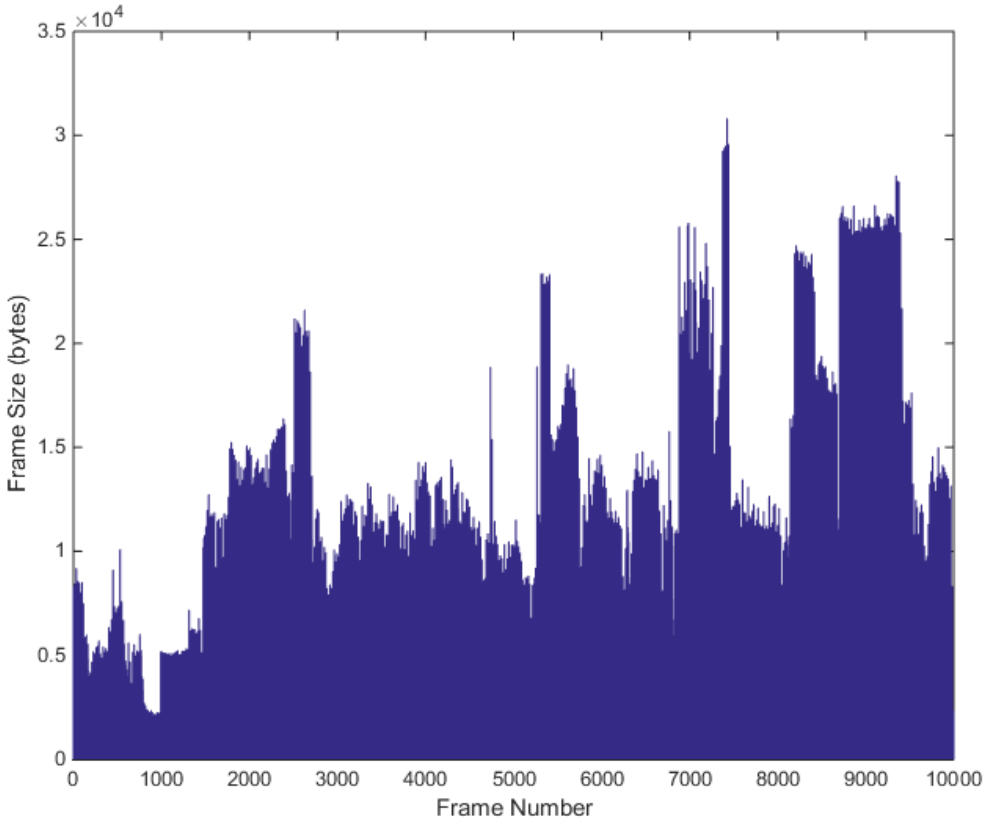


Figure 3.4 Frame Distribution Graphs for Movie Trace Files

3.4.2 Performance Measures

To evaluate the performance of the scheduling algorithms system throughput, fairness and BB frame loss ratio are used. The performance indexes employed are presented in the following.

1. **Throughput:** throughput is defined as the total number of bits transmitted over the total simulation duration. This means the system's average throughput is the sum of average throughputs over the total simulation duration, across all PLPs.

$$S_{throughput,i} = \frac{1}{N_{T2}} \sum_{j=1}^{N_{T2}} r_{i\,j} \qquad (3.23)$$

$$R^{av} = \frac{1}{N_{T2}} \sum_{j=1}^{N_{T2}} \sum_{i=1}^{N} r_{i\,j} \qquad (3.24)$$

$S_{throughput,i}$ is the average throughput of PLP i over the duration of the simulation, R^{av} is the average system throughput over the duration of the simulation and $r_{i\ j}$ is the throughput of PLP i on the j^{th} T2 frame.

2. **Fairness:** This is used to determine whether PLPs are receiving a fair share of resources. To measure fairness among PLPs, the Jain fairness index [35] is adopted as given below.

$$F = \frac{\left(\sum_{i=1}^{N} S_{throughput,i} \Big/ R_{throughput,i}\right)^2}{N \sum_{i=1}^{N} \left(S_{throughput,i} \Big/ R_{throughput,i}\right)^2} \quad (3.25)$$

$R_{throughput,i}$ is the actual average throughput requirement of PLP i over the duration of the simulation and is equal to average bit rate of the service carried by PLP i.

3. **BB Frame loss:** This determines the total number of BB frames that are dropped over simulation duration. BB frames may be dropped either due to buffer overflow and/or due to failure for PLPs to meet delay and data rate constraints when time slice allocation in employed as will be seen in chapter 5. It is assumed that all users have perfect channel conditions, therefore, there will be no BB frames lost due to poor channel conditions. The BB frame loss is give as a ratio between the dropped BB frames with respect to the BB frames that were sent. Video is very sensitive to packet loss, hence to BB frame loss. A high packet loss will damage the user experience. SVC is however more tolerant to packet loss because a loss of a packet from a higher layer will have a less dramatic effect on the video quality than a loss of a base layer packet. A lack of a return channel in DVB-T2 systems make retransmission of lost packets impossible. This makes an algorithm with a loss below the acceptable loss threshold favourable. For this research the acceptable loss threshold is arbitrarily set at 5% for all video services.

$$L_{BBframes} = \frac{\sum_{i=1}^{N} \sum_{t=1}^{N_{T2}} B_{discarded,i}(t)}{\sum_{i=1}^{N} \sum_{t=1}^{N_{T2}} B_{size,i}(t)} \quad (3.26)$$

Here$L_{BBframes}$ is the BB frame loss of the system, N_{T2} is the number of T2 frames over which the simulation is run, $B_{discarded,i}(t)$ is the total size of the discarded BB frames of service i at time t and $B_{size,i}(t)$ is the total size of all BB frames arriving into the buffer of service i at time t.

3.4.3 Algorithm for Developing Results

The algorithm aims to evaluate the results of the scheduling schemes implemented in this dissertation. It consists of obtaining the performance metrics presented in the previous subsection.

Table 3.6 Algorithm for Development of Results

Algorithm for Results Development
1: if service scheduling is present
2: Allocate bandwidth to SVC services according to the service scheduling scheme in Chapter 4.
3:end if
4:repeat
5: for each PLP
6: compute instantaneous bit rate
7: generate BB frames that correspond to the instantaneous bit rate and keep count of the generated BB frames
8: if (number of BB frame + buffer occupancy)>buffer size
9: drop the excess BB frames and keep count of the number of dropped BB frames.
10: end if
11: end for
12: Allocate BB frames to PLPs using the PLP scheduling schemes in Chapter 3.
13: if time slice allocation is present
14: Allocate time slices using the time slice allocation scheme in Chapter 5.
15: for each PLP
16: if not allocated time slices
17: drop BB frames and update count
18: end if
19: end for
20: end if
21: for each PLP
22: compute throughput from the allocated BB frames
23: end for
24:until all T2 frames are scheduled
25:compute performance metrics presented by equations (3.24), (325) and (3.26)

For each PLP, the algorithm computes instantaneous bit rates corresponding to each T2 frame. If there is service scheduling, the SVC services are allocated bandwidth, according to

the service scheduling scheme in Chapter 4, before instantaneous bit rates are computed. For each T2 frame, BB frames corresponding to an instantaneous bit rate are generated for each PLP and stored in a finite buffer. The algorithm keeps count of the generated BB frames. If the number of generated BB frames plus the ones remaining in the buffer exceeds the buffer size, the excess BB frames are dropped. The algorithm keeps count of the number of BB frames that are dropped. The PLPs are then allocated BB frames using PLP scheduling schemes presented in this chapter. If time slice allocation is present, PLPs are allocated time slices according to the time slice allocation scheme in Chapter 5. BB frames of PLPs that are not successful are dropped and the algorithm updates the count of number of BB frames that are dropped. The algorithm counts number of BB frames that are allocated and calculates throughput for each PLP. The process continues until all T2 frames are scheduled. After performing the scheduling, the algorithm computes the performance metrics values presented by equations (3.24), (3.25) and (3.26). The algorithm is summarized in Table 3.6

3.4.4 Simulation Results and Discussion

The simulation results and discussions of the achievable performance of the various schemes are also presented in this section. The performance of the scheduling algorithms with increasing number of services is analysed. The performance is evaluated using the performance indexes described in section 3.4.2. Video traffic model described in section 3.4.1 is used as input.

Figure 3.5 shows the throughput performance of the proposed schedulers and the reference scheduler. At lower loads (from 10 services to 14 services) the throughput performance is almost the same for all schedulers with the reference scheduler giving a slightly lower throughput than the other schedulers. As the number of services increase, MLWBOF and EXP/PF give the best throughput performance. This MLWBOF and EXP/PF perform allocation per BB frame and this ensures every BB frame is allocated thus achieving the maximum system capacity. The maximum capacity is reached at 15 services and is maintained as the number of services increase. The statistical multiplexers on the other hand perform a proportional sharing therefore some services may receive more BB frame than need. This results in some of the BB frames not being used and the maximum system capacity not being reached. MLWBOF and EXP/PF reach the maximum throughput at 15 services and maintains the same throughput as the number of services. The statistical multiplexers, on the other hand, show a drop in throughput as the number of services increase from 15 because the algorithms take a ceiling when determining the number of BB frame to

be allocated to each PLP, and this results in some BB frame not being allocated. As the number of services increase, the number of BB frames that are not allocated also increases and consequently throughput decreases. MLWBOF and EXP/PF are able to maintain the maximum throughput because they ensure that every BB frame is allocated. This, however, comes with some penalties such a higher complexity (quadratic) making them slower the statistical multiplexers which have a linear complexity. The scheduler either has to keep track of how many BB frames have been allocate to each PLP as it traverses all BB frames, or a way of counting the number of BB frames have been allocate to each PLP after all BB frames have been traversed.

MLWBOF and EXP/PF give a better BB frame loss performance. This is evident in Figure 3.6. This can be attributed to the fact that the schemes consider QoS demands in their respective algorithms. By considering buffer states, the algorithms always give priority to services with the highest buffer occupancies to avoid BB frame loss due to buffer overflow. Statmux, however, gives a slightly higher BB frame loss than other schedulers at lower loads i.e. from 11 services to 14 services, because it allocates resources less efficiently than the reference statistical multiplexer as it considers average bit rates instead of instantaneous bit rates in decision making. As the load increases from 14 services to 24 services, both statistical multiplexers give the worst loss performance because they do not prioritise buffer states in their algorithms. When traffic is low, there are enough data cells in a T2 frame to carry all the traffic but as the services increases the traffic exceed the amount of data cells available for data transmission. As a result, BB frame loss increases.

A fairness value of 1 is obtained when all services receive a fair share of resources. It can be observed from Figure 3.7 that schedulers obtain a fairness index performance close 1 for 10 services up to 12 services. The statistical multiplexers outperform the other schedulers with D-StatMux giving the best performance. D-StatMux maintains a fairness index that is greater than 0.95 while other schedulers experience a drop in fairness index as the number of services increases. By using instantaneous bit rate to determine the number of BB frame to be allocated to each PLP, D-StatMux gives each PLP a fair share of resources. MLWBOF and EXP/PF which consider QoS will always allocate resources services with high buffer occupancies leading to a high degree of unfairness. The statistical multiplexers on the other hand have a better fairness performance because they allocate resources proportionally between PLPs.

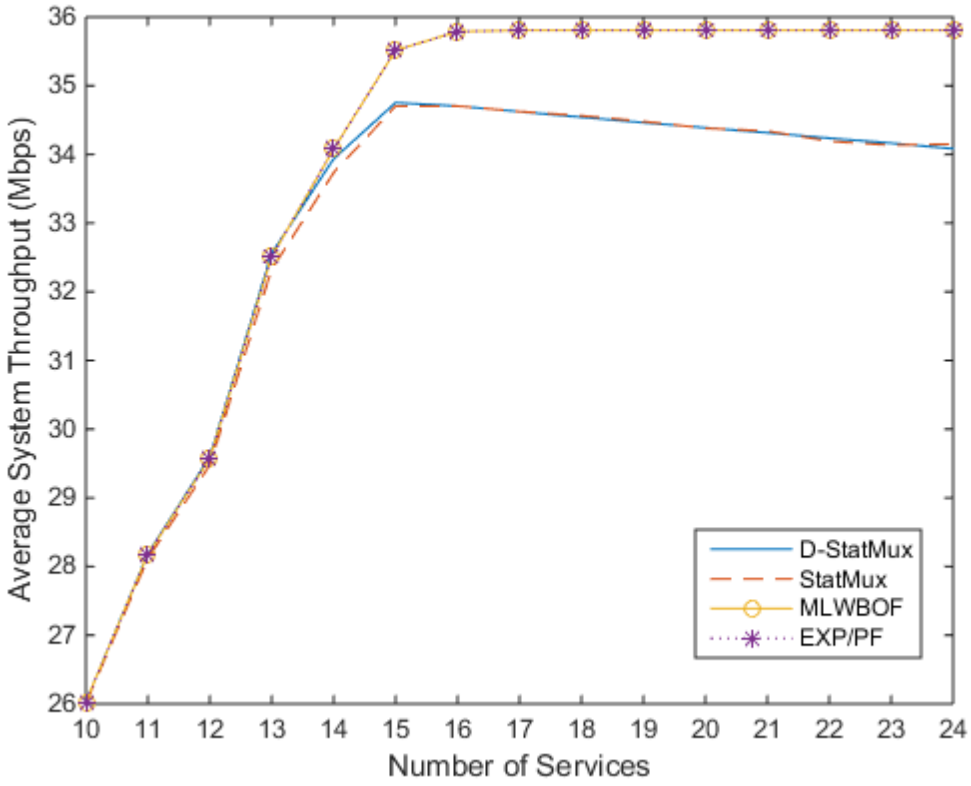


Figure 3.5 System Throughput with increasing number of services

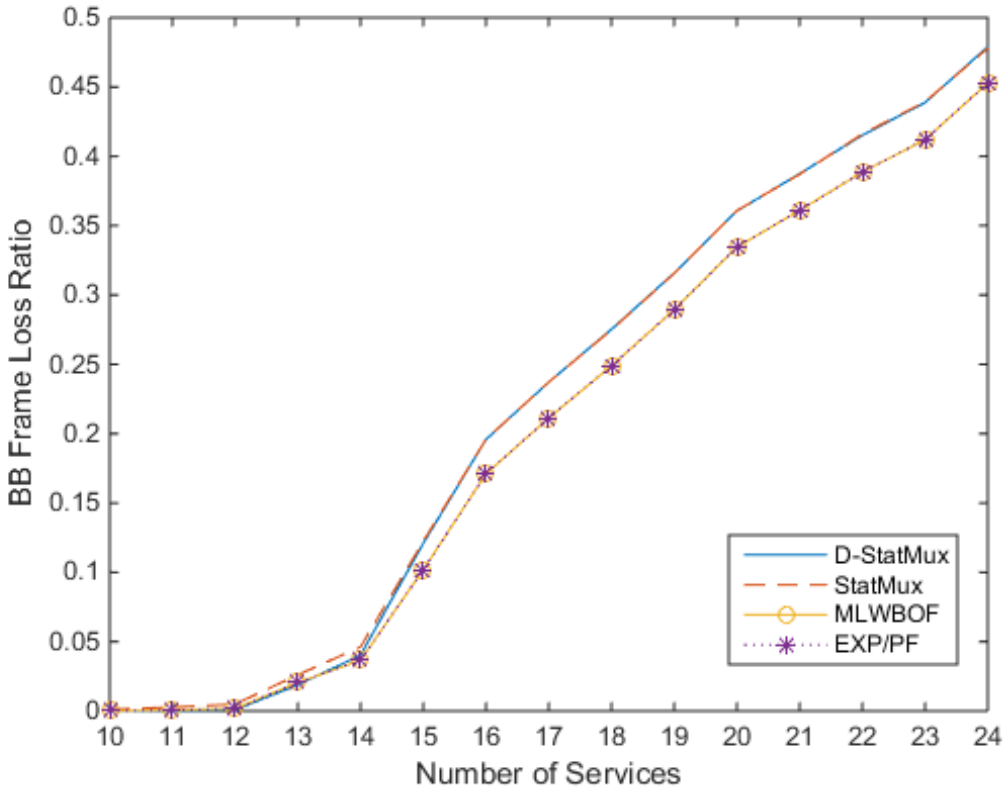


Figure 3.6 BB Frame loss with increasing number of services

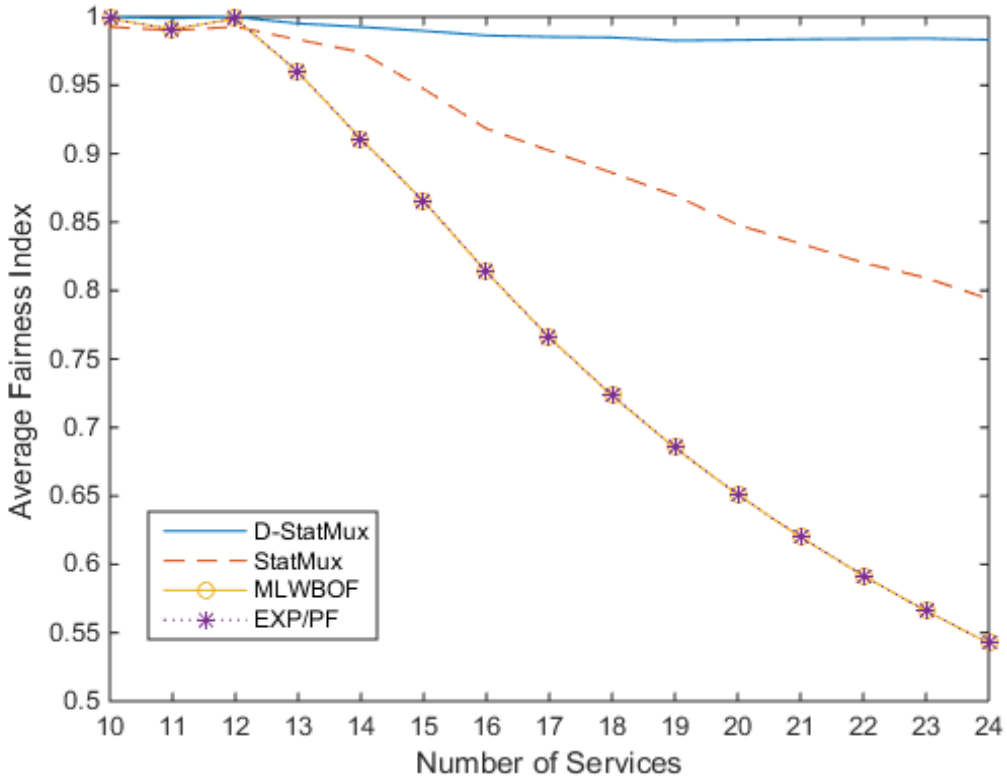


Figure 3.7 Fairness with increasing number of services

MLWBOF and EXP/PF give the same results as seen in the simulation results. In LTE the packet scheduling algorithms, MLWDF and EXP/PF, work on different classes of data with different delay and QoS requirements. For this research, the algorithms adopted for DVB-T2 work on only video traffic. Furthermore, MLWDF prefers users with the strongest acceptable loss and deadline expiration requirements while EXP/PF uses the exponential rule to give priority to the give priority to the service with the earliest deadline. Both algorithms use PF to achieve fairness. Since acceptable loss is the same for all services for this research, the proposed DVB-T2 schemes both prioritise buffer states and use PF for fairness, hence, they achieve the same results.

3.5 Chapter Summary

New PLP scheduling schemes, D-StatMux, MLWBOF and EXP/PF, are proposed in this chapter. A system model for PLP scheduling is described and elaborate description of the proposed scheduling schemes is presented. A simulation model which comprised of simulation parameters, a traffic model and DVB-T2 system parameters is described as well as the performance measures used to evaluate the performance of the proposed schedulers. Simulations are performed according to the described system model and performance is evaluated for an increasing number of services. The simulation results show that all the proposed schedulers perform better than the existing schedulers. From the results it can be seen that MLWBOF and EXP/PF give the best throughput and BB frame loss performance while D-StatMux outperform the other schedulers in terms fairness. A relationship between throughput and BB frame loss has been identified which is that schedulers that give the highest throughput have the lowest BB frame loss. This is because the higher the throughput the more BB frames are scheduled and the lower the number of BB frames that can be dropped. In general, the statistical multiplexers give the best fairness performance because they allocate resources proportionally to the services. MLWBOF and EXP/PF give the best BB frame loss performance because they consider buffer states when making scheduling decisions to avoid BB frame loss due to buffer overflow. The large drop in fairness of MLWBOF and EXP/PF as the number of services increases means the number of services that are satisfied also decrease. A mechanism that improves the fairness of the algorithms is therefore necessary. Although the throughput performance of the per-BB-frame schedulers is better than that of the statistical multiplexers at higher loads, the margin is very small. From

the simulation results, it can also be seen that as the number of services increase throughput and fairness decreases while BB frame loss increases. A mechanism that controls the BB frame loss as the number of SVC services increase is therefore necessary. It can be concluded that MLWBOF and EXP/PF give the best overall performance.

The statistical multiplexers, on the other hand, have a lower computational complexity (linear) than MLWBOF and EXP/PF which have a quadratic complexity. This implies that the statistical multiplexers are faster than MLWBOF and EXP/PF. However, all the algorithms can be easily implemented in real-time. A trade-off also has to be made between system performance and complexity when choosing a scheduling scheme. The next chapter proposes a service scheduling scheme that exploits the bit rate adaptability SVC by transmitting a subset of the video layers when the total traffic exceeds the channel capacity.

.

Chapter 4 Joint PLP and Service Scheduling

4.1 Introduction

One of the goals of DVB-T2 is to deliver services for fixed and portable receivers and possibly to mobile reception. This means different versions of the same content is encoded separately for each reception scenario and transmitted. SVC, which was developed as an alternative to simulcast, provides different versions in a single video stream by encoding the video stream in multiple layers and each receiver can decode only the layers it is capable of receiving. The input pre-processing (IPP) block in the DVB-T2 architecture (Chapter 2, Section 2) may assume the role of the scheduler to prepare services for T2 processing. In this dissertation a service scheduler that exploits the bit rate adaptability of SVC services to allocate bandwidth to SVC video services is proposed and implemented at the IPP.

The service scheduling is adopted from SVC video transmission in LTE multicasting [36-39]. The schemes exploit scalability and ensure the base layer is received by all multicast groups and enhancement layers are allocated according to factors such as price paid for a service, spectral efficiency and the number of users that will be served. The price based scheme described in [37] defines three classes where different class users pay different prices and enjoy different QoS. More quality layers are allocated to users who pay a higher price. In [36, 38, 39] subgrouping and frequency selectivity are used. A multicast group is split into subgroups and constrains transmission rate to the user experiencing the worst channel state. Frequency selectivity is achieved by assigning a frequency resource to the subgroup that guarantees the highest value of a defined cost in order to minimise the number of resources allocated for transmission. All the algorithms make use of channel state information (CSI) when selecting which multicast group to service and provide unequal error protection to different layers of an SVC bit stream. However, due to the lack of a return channel and long frame durations, which make it difficult to predict CSI even if a return channel was present, DVB-T2 does not use CSI to perform scheduling decisions. The resource allocation strategy defined in [38] ensures that all users receive at least the base layer of an SVC stream. Base layer allocation and enhancement layer allocation are performed separately beginning with base layer allocation. The enhancement layer allocation is performed when all multicast groups have been allocated a base layer and enhancement layers are allocated to subgroups which are capable of receiving the higher layers.

The service scheduling proposed in this study is adopted from [38]. The PLP scheduling in chapter 3 is combined with the service scheduling and performance is evaluated for the proposed PLP scheduling algorithms. In Section 4.2 a system model for PLP scheduling combined with service scheduling is described. Section 4.3 describes the proposed service scheduling scheme. The simulation results are provided and analyzed in Section 4.4 and Section 4.5 summarizes and concludes the chapter.

4.2 System Model

4.2.1 System Model Description

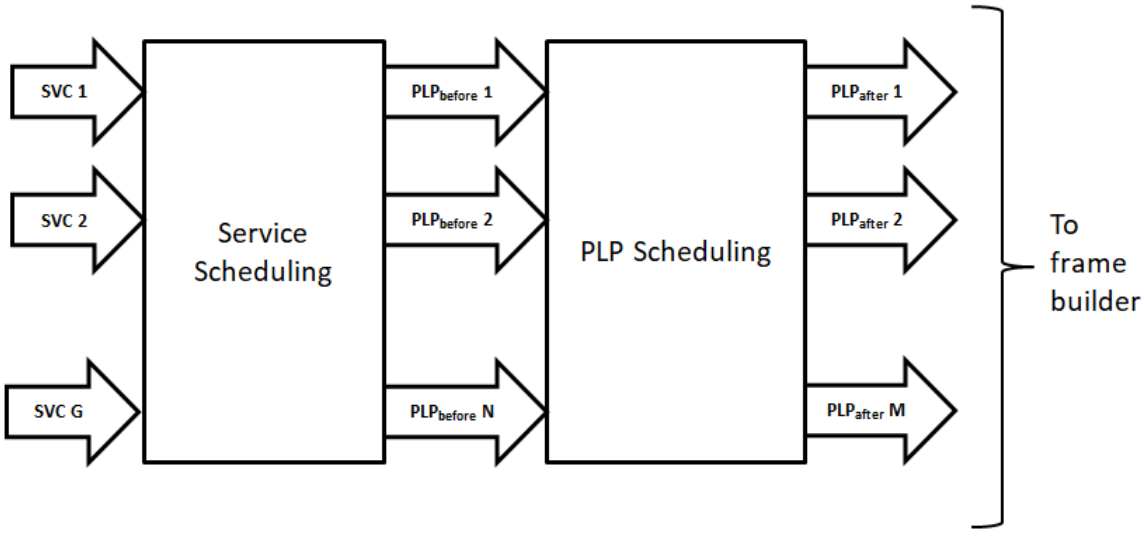


Figure 4.1 System Model for Joint Service and PLP Scheduling

Let G be the set of scalable videos each encoded into L layers which consist of a base layer and $L-1$ enhancement layers. $l=0$ indicates the base layer, $l=1$ is the first enhancement layer and so on. The base layer which provides basic quality and is essential for decoding the whole video frame is transmitted for every video. All layers of a video are mapped to the same PLP. The input bit rate is C_{input} and the total bit rate of the video layers selected for transmission must not exceed this value. The set of resources units in a frame i.e. BB frames, is denoted by B_{T2}. H.264/AVC and SVC provide low average bit rates but high traffic variability therefore the difference between the average and the peak bit rate is very large

[23]. This poses a challenge for efficient scheduling. To accommodate the high traffic variability and hence minimize BB frame loss, the proposed service scheduling multiplexes the video services based on their peak bit rate. However, provision of capacity based on peak bit rate results in inefficient use of bandwidth. Therefore, the peak bit rate is divided by 2. Let $r_{g,l}$ denotes half the peak bit rate and $R_{g,l}$ the average bit rate of the lth layer relevant to video g. The sum of the average bit rates of the layers selected for transmission for video g will be denoted by R_g which can be expressed as

$$R_g = \sum_i^L R_{g,i}. \quad (4.1)$$

$R_{g,l}$, R_g and the peak bit rates are obtained during source coding and compression. Finally $b_i \subseteq B_{T2}$ represents the number of BB selected for transmission of PLP $i, i = 1,2, \dots, N$ carrying such a video.G is the number of services being transmitted, N is the number of configurable PLPs and M is the number of PLPs that have been allocated BB frames. The system model is depicted in Figure 4.1. For G SVC service encoded with L layers, the service scheduling allocates bandwidth, C_{input}, to their base layers and then allocates bandwidth to enhancement layers of services whose base layers have been scheduled. The scheduling is performed using to predefined policies and continues until all services have been allocated or the bandwidth is exhausted. The SVC services are allocated bandwidth according to the input bit rate and the service scheduled for transmission are mapped to N PLPs and allocated BB frames using the PLP scheduling algorithms described in Chapter 3.

4.2.2 System Constraints

The following constraints are imposed:

a) The total bit rate of the videos selected shall not exceed the input capacity.

b) A PLP shall be assigned to one video.

c) A video shall be assigned one PLP.

d) The base layer shall be delivered for all videos

e) A video shall be assigned resources for a given layer only if previous layers have been allocated

Cost Function

In [38] two cost functions are defined to guarantee high spectral efficiency while assuring fairness. They are based on the ratio between the number of users in a subgroup and the number of resource requested by the group. The first cost function accounts for fairness through a logarithmic ratio between the obtained data rates and maximum data rate values. The second cost function considers fairness through the ratio between the total number of video layers and the index of the next video layer to be scheduled. The second cost function has been adopted in this dissertation and only aims to ensure fairness. The cost function g^* gives priority to videos that still miss a great number of layers compared to others. The cost function can be written as

$$g^* = arg\,max_{g \in G} \frac{L_g}{l_g} \quad (4.2)$$

where L_g is the total number of layers of video g and l_g is the index of the next layer to be scheduled. It is assumed that the encoding parameters, such as number of video layers and the average bit rate and peak bit rate of each layer of each SVC video service are known. For solutions that achieve the same maximum cost value, the solution requiring the least resources is chosen.

4.3 Service Scheduling

Service scheduling comprises the base layer and enhancement layer allocations. The LTE scheme in [38] provides each multicast group with resource needed to deliver the base layer. The quality of the video is increased by allocating resources for enhancement layers depending on the channel conditions and available resources.

The base layer allocation algorithm computes the modulation and coding scheme (MCS) on each resource block for each multicast group. The aim is to exploit frequency selectivity. The MCS is selected such that it supports all the users in the multicast group i.e. even the ones with poor channel conditions. Base layer assignment begins with computing the number of resources needed to deliver the base layer for each group and the minimum resource will be those associated with the highest MCS. If the available resource cannot guarantee delivery of the base layer to such a multicast group the group is removed from the list. The remaining groups are then allocated resource beginning with the group that requires the least resources. If the groups require the same resource, one that serves the most users is selected. This aims

to minimise resource consumption and maximise capacity by serving the highest number of users possible. Once a group is allocated resources it is removed from the list and the resources are updated. The algorithm stops when all groups have been served or the resources are exhausted.

For all the groups that have been allocated base layers subgroups are formed for enhancement layer allocation. For each member of a multicast group, a subgroup contain user's that experience a mean channel quality greater or equal that of such a member. Once the subgroups have been defined the sustainable MCS for available resource blocks are evaluated. The algorithm then determines the most the suitable resource blocks for the delivery of the required video layer to each subgroup. If the available resources cannot guarantee the transmission of the considered layer for any subgroup, the multicast group is removed from the list. The best group is selected from the remaining multicast groups using the cost functions described in Sub-section 4.2.2. The best group is one that guarantees high intra-spectral efficiency while assuring fairness. Once a group is selected, resource blocks are allocated to it and the available resource blocks are updated and the video layer is marked as scheduled. A group is removed from the list if all its enhancement layers have been assigned. The iteration stops either when no more resource blocks are available or when all the video layers have been transmitted to all multicast groups.

This base layer and enhancement layer allocation algorithms have been adopted and modified for DVB-T2 as described in Sub-sections 4.3.1 and 4.3.2. Since there is no knowledge of CSI, the algorithms in DVB-T2 only aim at sharing the bandwidth between video services in a fair manner. In the proposed scheme, the base layer allocation is performed first to ensure that at least the basic video quality is transmitted for each service. The enhancement layer allocation follows when there is capacity remaining to improve the video quality of the services. The scheduling algorithms are described as follows.

4.3.1 Base Layer Allocation

The base layer allocation is summarized in Table 4.1. The algorithm sorts the services in increasing order of $r_{g,0}$ and allocated capacity starting from the front of the queue i.e. with the service that requires the least amount of capacity. For each SVC video, the algorithm sets the video layer to be allocated bandwidth as the base layer i.e. layer 0 (line 1-3). If several services require the same amount of resources for the base layer the service is selected randomly. Line 5 indicates that video which requires the least resources is selected. $r_{g,0}$ is

half the peak bit rate of the base layer. In a case that the resources cannot guarantee the base layer transmission, the video is removed from the G set. The solution tries to minimise the number of services that are not served when the available capacity is not enough to allocate all scheduled services. Once a video is selected the available capacity is updated and the index of the next video layers to be allocated bandwidth is set to 1. The iteration continues until all the services are served or no more capacity is available.

Table 4.1 Base layer Allocation Algorithm

Base Layer Allocation Algorithm
1: for all $g \in G$ ***do***
2: $l_g = 0$
3: end for
4: repeat
5: ***Select*** $g^* = arg\,min_{g \in G \mid l_g = 0} \left| r_{g,0} \right|$
6: ***if*** $r_{g^*,0} > C_{input}$ ***then***
7: ***Update*** $G = G - \{g\}$
8: ***else***
9: $l_{g^*} = 1$
10: ***update*** $C_{input} = C_{input} - r_{g^*,0}$
11: until $\left\{ g \in G \mid l_g = 0 \right\} = 0 \ \vee \left| C_{input} \right| = 0$

4.3.2 Enhancement Layer Allocation

The enhancement layer allocation is summarized in Table 4.2. G now represents the set of SVC services whose base layers have been assigned and C_{input} represents the bandwidth available after base layer allocation. Lines 1-7 determine the videos which are eligible for enhancement allocation. The services that have been allocated base layers but have only one layer are removed from G. The scheduled service is then selected for the current iteration (line 9). The selection is performed according to a cost function expressed in equation (4.2) in order to ensure fairness. For the services with the same cost, the service which requires the least capacity is selected. Once a service is selected, the available bandwidth is updated and l_{g^*} is updated i.e. the current layer is marked as scheduled. A service g^* is deleted from the G set when all its enhancement layers have been allocated bandwidth. The iteration goes on

until there is no more bandwidth available or all the layers of each service have been allocated.

Table 4.2 Enhancement layer Allocation Algorithm

Enhancement Layer Allocation Algorithm
1: for all $g \in G$ ***do***
2: ***if*** $L_g > 1$ ***then***
3: $l_g = 1$
4: ***else***
5: ***Update*** $G = G - \{g\}$
6: ***end if***
7: end for
8: repeat
9: Perform video selection using (4.2)
10: $l_{g^*} = l_{g^*} + 1$
11: if $l_{g^*} > L_g - 1$ ***then***
12: ***Update*** $G = G - \{g^*\}$
13: end if
14: Update $C_{input} = C_{input} - r_{g^*, l_{g^*}}$
15: until $\lvert G \rvert = 0 \vee \lvert C_{input} \rvert = 0$

4.4 Computational Analysis

The computational complexity of the service scheduling algorithm is computed in this chapter. First, the computational complexity of the base layer allocation is determined. The computational complexity of the enhancement layer allocation algorithm follows. These are then used to determine the overall complexity of the service scheduling scheme.

The complexity of the code in lines 1-3 of the base layer algorithm is $O(G)$ and the complexity of the code in lines 4-11 is $O(GC_{input})$. The overall complexity for base layer allocation is $O(G + GC_{input})$.

The complexity of the code in lines 1-7 of the enhancement layer allocation is $O(G)$. The complexity of the code in lines 8-15 is $O(GC_{input}L)$ where L is the maximum number of layers to be transmitted, and the complexity of code in line 9 is $O(G)$. The overall complexity for enhancement layer allocation is $O(G + G^2C_{input}L)$.

The overall complexity of the proposed service scheduling scheme is $O(G + GC_{input} + G + G^2C_{input}L) = O(2G + GC_{input} + G^2C_{input}L)$. The scheme has a quadratic complexity.

4.5 Simulations

The simulations are run for different values of alpha, α, ranging between 1 and 4 and the allowable input capacity, C_{input}, is given by

$$C_{input} = \alpha \times C_{channel} \quad (4.3)$$

$C_{channel}$ is the channel capacity for the specified physical layer configurations. Equation (4.2) means the proposed system allows more input bandwidth that the available channel bandwidth and this is called channel over-packing. Over-packing exploits the fact that service scheduling admits services according to mean bit rate values while video bit rates are variable in nature and the services have statistically varying bit rates. This way more services, in this case video layers, can be scheduled. Simulation results are generated for an increasing number of services when $\alpha = 1$ and $\alpha = 3$. The same simulation and traffic model (see Section 3.4.1 and Section 3.4.2) used for the PLP scheduling in Chapter 3 is used to perform the simulations in this chapter. Performance is evaluated for the PLP schedulers presented in chapter 3 namely D-StatMux, MLWBOF and EXP/PF. Because MLWBOF and EXP/PF give the same performance, MLWBOF is considered in the simulations to present the performance of the adopted schemes. The evaluation is performed against StatMux for the case where no service scheduling is used.

The simulation results in Figure 4.2 show that service scheduling results in poor throughput performance at lower values of α. This is because service scheduling only selects a subset of layers for transmission which results in low traffic levels as opposed to transmitting all layers of each service as is the case when there is no service scheduling. Increasing the value of α results in an increase in traffic because more services and/or video layers are scheduled for transmission. Consequently, throughput increases. When α=1, video base layers and some enhancement layers are transmitted when there are 3 and 6 services resulting in an increase in throughput. As the services increase, only base layers are scheduled and this results in a drop in throughput as seen when services increase from 6 to 9. The throughput remains almost

constant as the number of services increase because the input threshold is reached. It can be seen from the throughput results that when $\alpha = 3$ the throughput performance of the schedulers is lower than the performance when there is no service scheduling by a small margin.

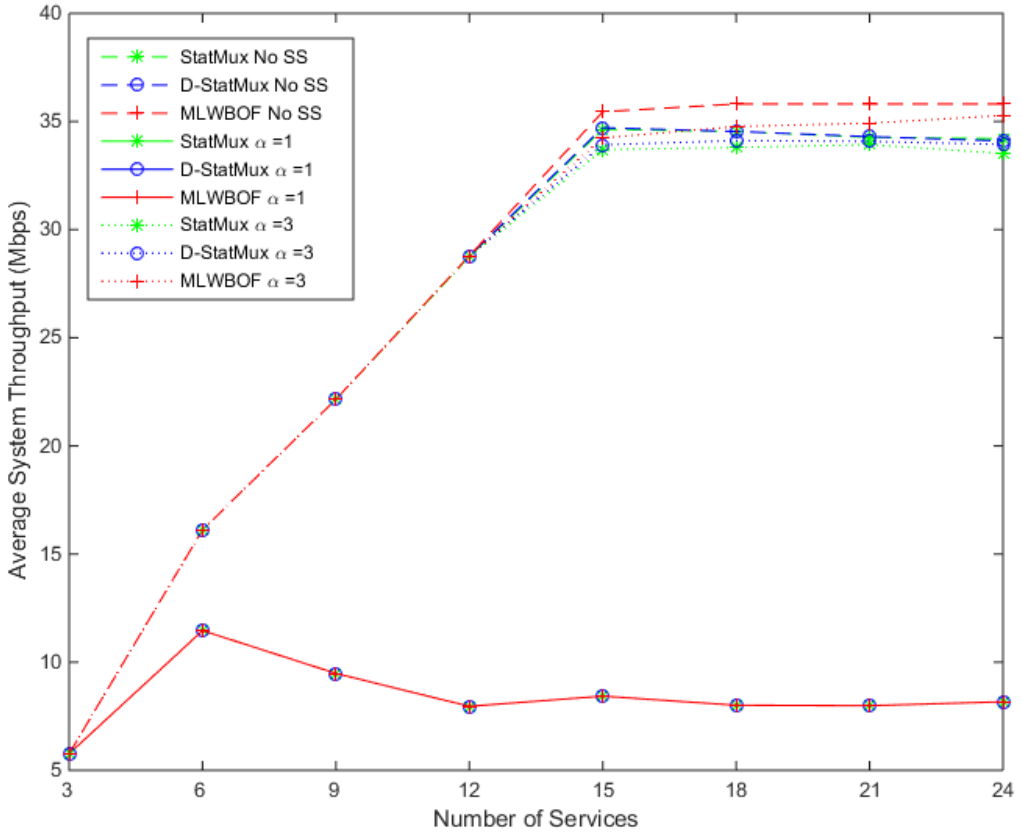


Figure 4.2 System Throughput with increasing number of services

Similar to the case of throughput, service scheduling results in a low BB frame loss at lower values of α as seen in Figure 4.3. This because there is less traffic to be transmitted and there are enough data cell in a T2 frame to carry all the traffic. Hence, at $\alpha = 1$ there is no BB frame loss. As α increases traffic increases and BB frame loss increases as seen when $\alpha = 3$. From 3 to 12 services, all the scheduled traffic is able to be transmitted with almost no loss. When the number of services increases, the amount of traffic increase and the T2 frames cannot carry all the traffic. As a result there is a loss of BB frames in the system. At $\alpha = 3$, the system is able to maintain a loss below the acceptable loss threshold (5%) for up to 21 services, with a system throughput above 30Mbps. Without service scheduling, the loss kept

below 5% for up to only 12 services with a system throughput below 30Mbps. Thus, service scheduling results in a better loss performance.

Figure 4.3 displays the fairness index performance of the PLP schedulers. With service scheduling the fairness index is almost 1 for all schedulers when $\alpha = 1$. This is because the data rate requirements of all the scheduled services are satisfied. When $\alpha = 3$, the fairness indexes of the schedulers decrease with MLWBOF still giving the worst performance. This is because as α increase traffic also increases and MLWBOF prioritises buffer states to reduce BB frame loss over fairness while the statistical multiplexers allocate resources proportionally thus ensuring fairness. At this value of α, the statistical multiplexers are able to maintain a fairness index greater than 0.95 with service scheduling, while MLWBOF gives worse performance than when there is no service scheduling. Therefore, service scheduling improves the fairness performance of statistical multiplexers.

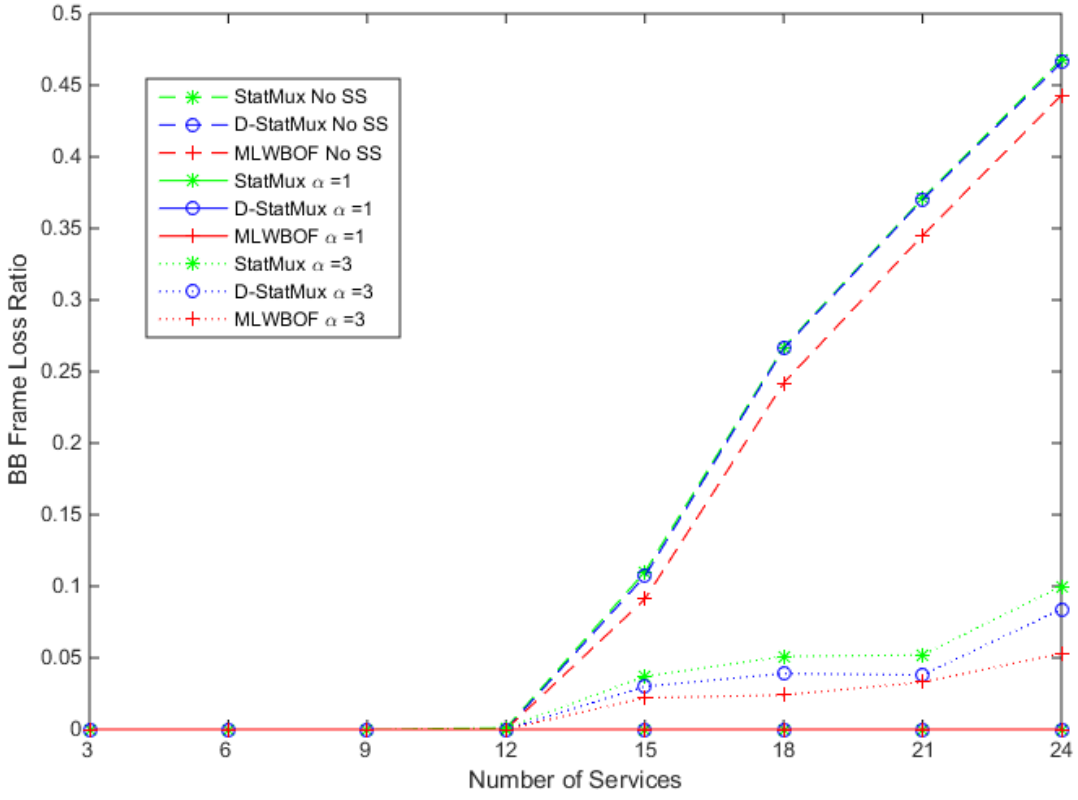


Figure 4.3 BB Frame loss with increasing number of services

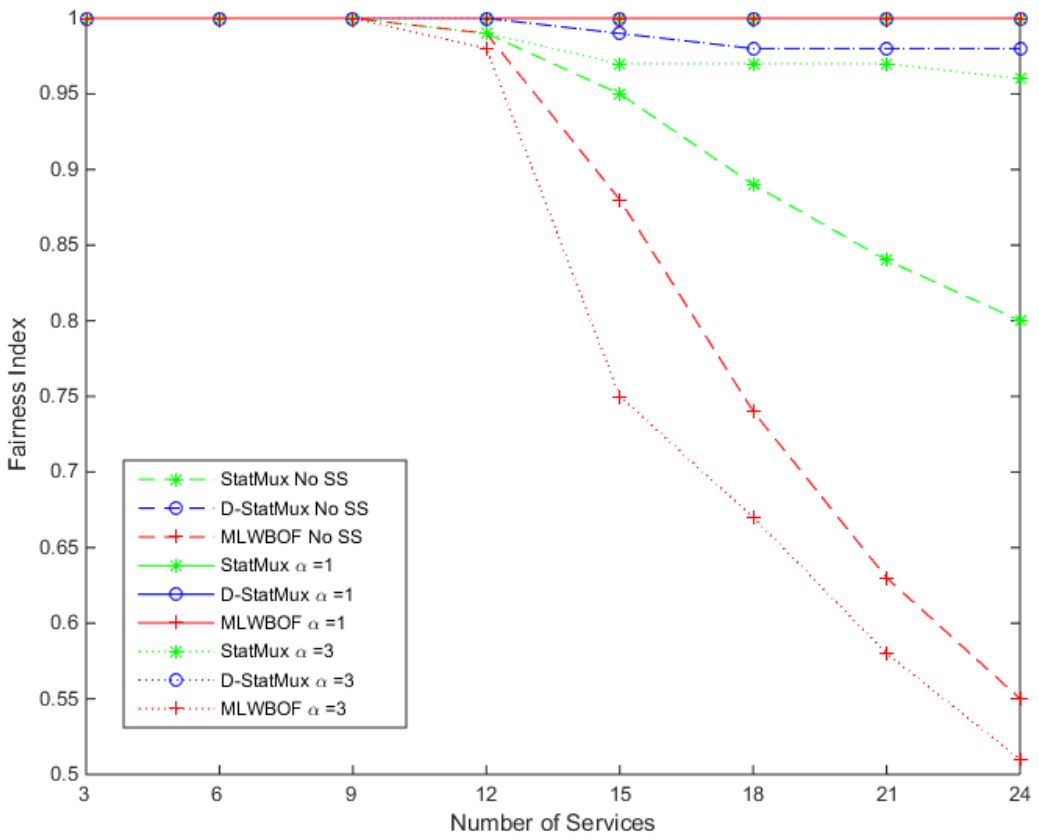


Figure 4.4 Fairness with increasing number of services

The simulation results show that service scheduling reduces BB frame loss but compromises on throughput, and fairness for MLWBOF. As α increases, throughput increases and loss increases as seen when $\alpha = 1$ and when $\alpha = 3$. It is therefore necessary to find a value of α that reduces BB frame loss but achieves a reasonable throughput i.e. a value that makes a good trade-off between throughput and BB frame loss without compromising much on fairness.

4.6 Chapter Summary

A service scheduling scheme for SVC services has been proposed in this chapter. The scheme comprises of a base layer and enhancement layer allocation and aims to provide at least the basic video quality for each service while allocating bandwidth efficiently and fairly. The allocation algorithms are described elaborately in this chapter. A system model is described for simulation. The same simulation model and performance measures used in chapter 3 are used for the simulations and performance evaluation respectively. The performance of the

PLP schedulers proposed in chapter 3 is evaluated according to the described system model. It is evaluated for an increasing value of α and an increasing number of services. The simulation results are compared to the results in chapter 3 where there is no service scheduling. According to the simulation results, the service scheduling results in an improved overall performance at an optimal value of α. The schedulers give similar performance as in chapter 3. MLWBOF still gives the best throughput and BB frame loss performance because they ensure that every BB frame is allocated and consider buffer occupancies in decision making while D-StatMux outperform the other schedulers in terms fairness because it shares resources more proportionally between PLPs. Through service scheduling a good trade-off between throughput and BB frame loss has been made without compromising much on fairness, with the exception of the per BB frame schedulers. However, some services are delay-sensitive and the schedulers proposed so far do not guarantee timely delivery of such services. In the next chapter, a time slice allocation scheme proposed in [24] is incorporated in the system model and the performance of the PLP scheduler is evaluated.

Chapter 5 Joint Service and PLP scheduling with Time Slice Allocation

5.1 Introduction

In communication systems timely delivery of services is crucial. Some services have strict delay constraints and failure to meet those delay constraints results in loss of service in that instant in time. The current time slice allocation defined for DVB-T2 does not consider services delay constraints and consequently the delay constraints are not met satisfactorily. This deteriorates the performance of the system due to packet loss and ultimately the throughput performance of the system. In this chapter the time slice allocation proposed in [24] is combined with PLP and service scheduling schemes proposed in Chapters 3 and 4 respectively. The performance of the PLP scheduling schemes is evaluated for the hybrid system. In Section 5.2 a system model for Service and PLP scheduling with time slice allocation is described. Section 5.3 describes the time slice allocation scheme and the simulation results are provided and discussed in Section 5.4. Section 5.5 summarizes and concludes the chapter.

5.2 System Model

5.2.1 System Model Description

The system model is as depicted in Figure 5.1. Let G be the number of scalable videos. All layers of a scalable video are mapped to one PLP. The service scheduling must be performed in such a way that the total bit rate of the layers selected for transmission does not exceed the allowable input bit rate. Each T2 frame consists of BB frames denoted by B_{T2}. A PLP is allocated $b_i \subseteq B_{T2}$ BB frames for transmission where i is the index of a PLP. N is the number of PLPs carrying video services and M is the number of services selected for transmission. Similar to [24], the data cells of a T2 frame are organised into equal sized data slices and the number of slices available is given by

$$N_s = \left\lfloor \frac{C_{data} L_{data}}{K} \right\rfloor \quad (5.1)$$

where K is the number of data cells in a slice, C_{data} is the number of data cells in a data symbol and L_{data} is the number of data symbols. Figure 5.2 illustrates the proposed data

slices in a T2 frame. Each PLP is associated with a weight W_i, a required number of data slices S_i and a delay constraint D_i. Let the address of the last cell a PLP can be mapped to in order to satisfy D_i be denoted by T_i. The relationship between D_i and T_i is given by

$$T_i = \left\lfloor \frac{(D_i - T_P - T_s N_{P2}) C_{data}}{T_s K} \right\rfloor - 1 \qquad (5.2)$$

where K is the number of data cells in a data slice, T_s the duration of a data symbol, N_{P2} is the number of P2 symbols and C_{data} is the number of data cells in a data symbol. The SVC services are allocated bandwidth according to the service scheduling proposed in Chapter 4 and the scheduled services are mapped to PLPs. The PLPs are then allocated BB frames using PLP scheduling algorithms proposed in Chapter 3 following which the PLPs are allocated time slices using the time slice allocation scheme proposed in [24]. The time slice allocation scheme is described in the next section. The time slice allocation scheme aims to distribute time slices to M PLPs that have been allocated BB frames by the PLP scheduler based on their data rate and delay requirements. The scheme drops BB frames of PLPs whose data rate and delay requirements are not satisfied and admits Q PLPs who are allocated time slices successfully.

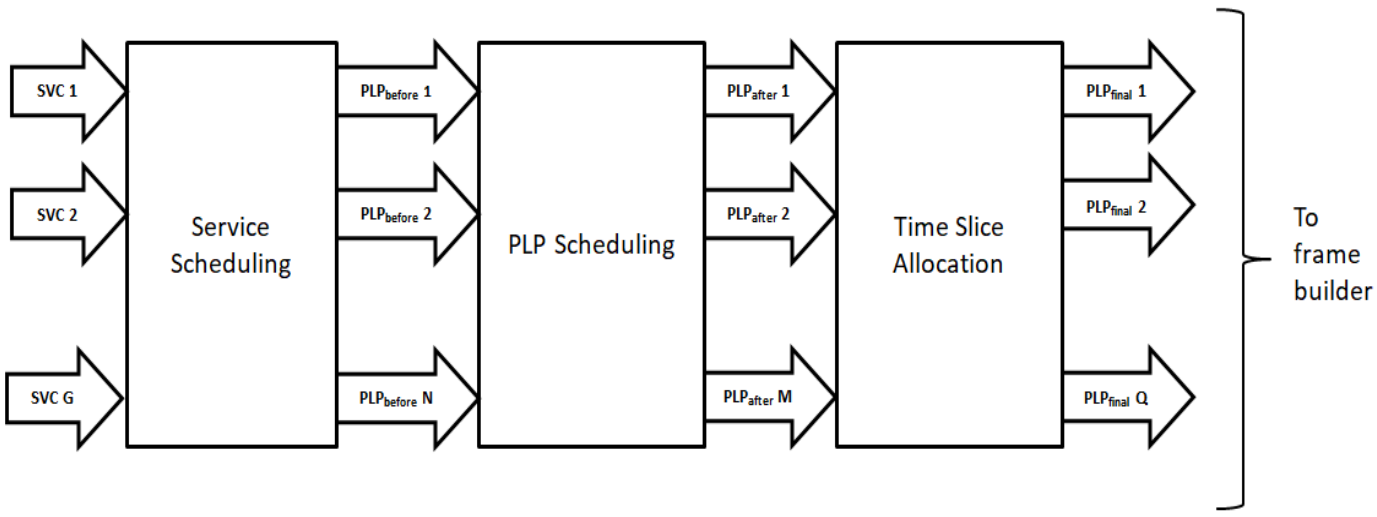


Figure 5.1 System Model for Joint Service and PLP Scheduling with Time Slice Allocation

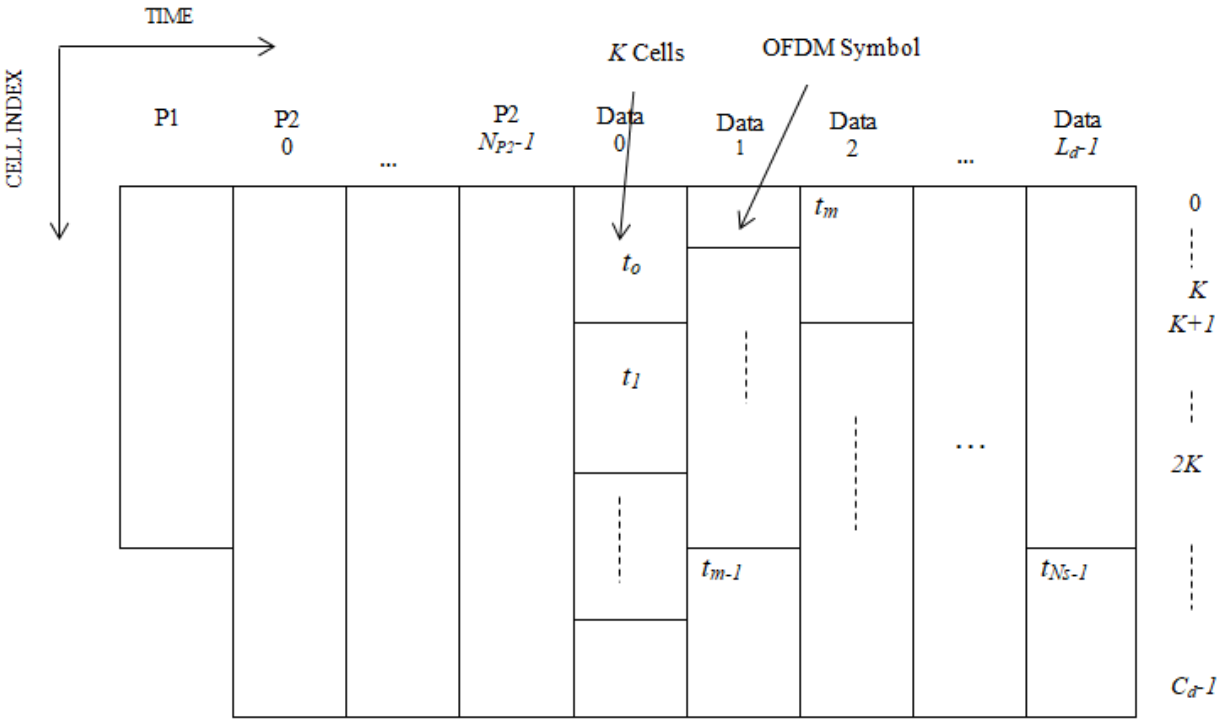


Figure 5.2 Slice Allocation Scheme's proposed data slices in a T2 Frame

5.2.2 Time Slice Allocation Constraints

The constraints imposed are:

a) A data slice can be allocated to one and only one PLP.
b) A PLP cannot be allocated more slices than it needs

5.3 Time Slice Allocation

The lack of a return channel and long transmission intervals in DVB-T2 make it necessary for an efficient resource allocation scheme that considers service delay constraints to ensure timely delivery of those services. In current slice allocation procedures defined by DVB-T2 critical services may be delivered as Type 1 in order to meet delay requirements but this scheme is unreliable because it does not provide diversity against time or frequency selective fading. For this reason a reliable and efficient slice allocation scheme which meets data rate and delay requirements while providing diversity capability is proposed in [24]. Unlike Type 2 PLPs which contain an equal number of sub-slices and different number of cells per sub-

slice, the scheme divides the cells of a T2 frame into equal sized data slices, encodes them in binary numbers and organizes the data slices in a binary tree. Different PLPs may be mapped to different number of sub-slices according to their data rate requirement. This may however introduce additional signalling cost to the system since signalling must indicate the address of data slices. In order to solve this problem the interval, I_s, between consecutive data slices allocated to a PLP is made to be equal. The interval is set to a power of 2 and depends on the level, l_{node}, of the node on the slice allocation tree as in Equation (5.3)

$$I_s = 2^{l_{node}} \quad (5.3)$$

Therefore, if t_{start} is the address of the first sub-slice then the address of the j^{th} sub-slice is given by

$$t_j = t_{start} + (j-1)I_s \quad (5.4)$$

A one-dimensional addressing scheme where $a_{l,c} = a(n)$ if and only if $n = C_d l + c$ is used. l ($l = 0,1,2,\dots, L_{data} - 1$) is the index of the data symbol and c ($c = 0,1,2,\dots, C_{data} - 1$) is the index of the cell in a data symbol. The addressing scheme is depicted in Figure 5.2. When a receiver receives cells of a PLP it does not decode them until all the cells of a TIB are received. Therefore the position of the last cell a PLP is mapped to can be determined from the delay. The delay is given as

$$d_i = T_P + T_s\left(N_{P2} + \left\lceil \frac{n_i+1}{c_d} \right\rceil\right) \quad (5.5)$$

where $\lceil \bullet \rceil$ denotes the integer ceiling operator. n_i is the address of the last cell of the slice allocated to PLP i and can be found by rearranging equation .

The scheme organizes the data slices in a complete binary tree. A complete binary tree is a tree that is completely filled, with the possible exception of the bottom level and the bottom level is filled from left to right. The structure of the slice allocation tree is as depicted in Figure 5.3. The data slices make up the leaf nodes of the tree. The height of such a tree is

$$h = \lceil log_2 N_{nodes} \rceil. \quad (5.6)$$

where h is the height of the tree and N_{nodes} is the number of leaf nodes in the tree. The slice allocation is therefore carried out with the aid of a complete binary tree of height L where

$$L = \lceil log_2 N_s \rceil. \quad (5.7)$$

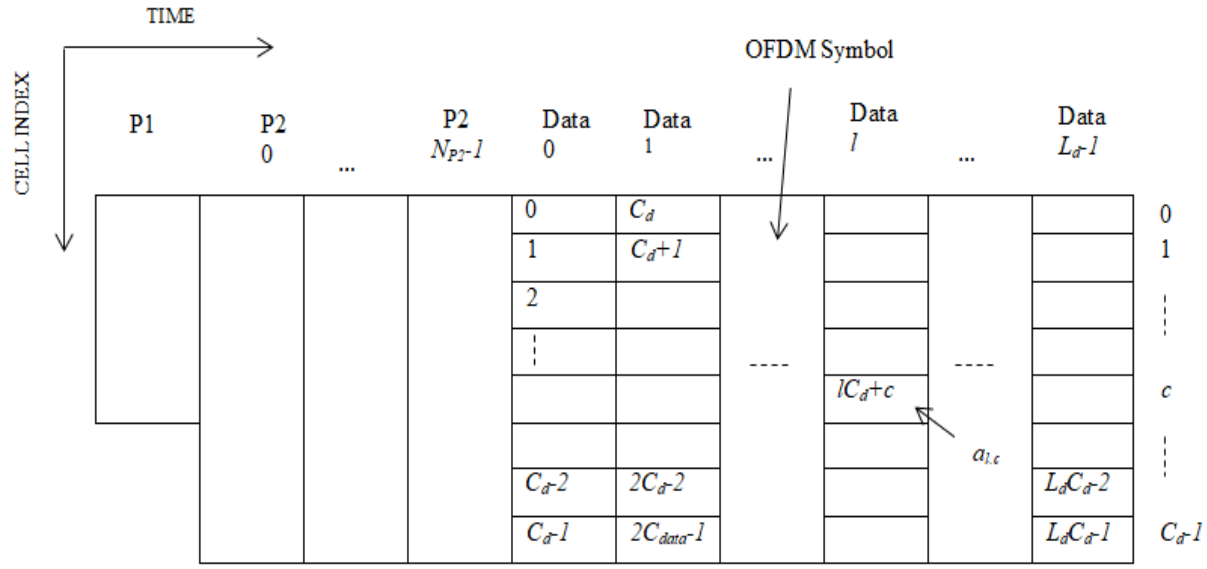


Figure 5.3 Addressing Scheme

The data slices are allocated to each node such that it has non-adjacent slices. The tree root is at level 0 while the child nodes are on level 1 and so on. Each node is associated with a parent node, left and right child nodes, a level, data slices, the quantity of data slices, a label and a state. The nodes are described using the following structure

$$[str = \{Parent, Left, Right, Level, Label, AvaNum, Content, State\}]$$

The length of the label is equal to the level and the first digit is either 0 or 1 and the rest of the digits of the node label are the same as the parent label. The state is equal to one when the interval between any adjacent data slice is the same; otherwise it is zero. A service whose data rate and delay requirements are not met is not allocated any slices in the current frame to increase chances of meeting requirements for remaining services.

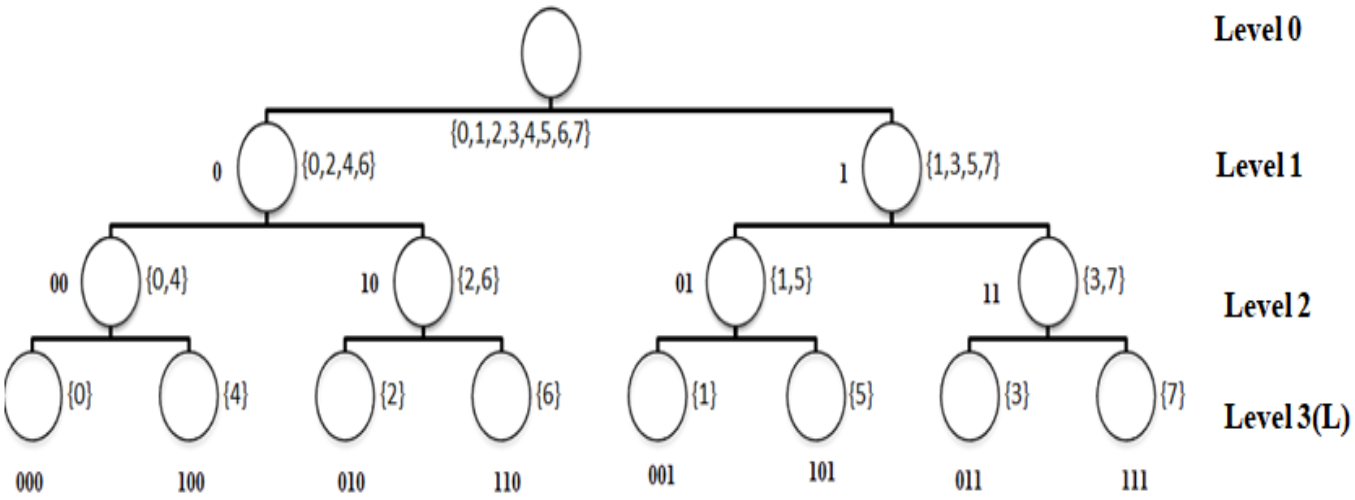


Figure 5.4 Slice Allocation Tree Illustration

The slice allocation is described in two algorithms: a PLP slice allocation algorithm that describes how one PLP is allocated data slices and a heuristic algorithm that puts PLPs in a queue using predefined policies and allocates data slices to PLPs according to the PLP slice allocation algorithm. The slice allocation process is summarized as follows.

1) The PLPs are put in a queue in descending order of their priorities. The priority of a PLP is determined as follows: a PLP with the earliest last cell address, i.e. stricter delay constraint, has higher priority. If two or more PLPs have the same last cell address the PLP with the higher weight is preferred. The weight is a function of the delay constraint and the smaller the delay constraint, the larger the weight. If two or more PLPs have the same weight, the PLP that requires more data slices has higher priority.
2) All nodes that meet the data requirement and delay constraint of a PLP, whose interval between any adjacent data slice is the same and whose children do not meet the data rate requirement of the PLP are searched for. From these nodes, the most suitable node is found using the following criteria: the node with the least number of slices is preferred and if two or more nodes have the same number of slices the node at a higher level of the tree is preferred. If two or more nodes are at the same level the node whose first slice is the earliest is preferred.

3) Allocated slices are removed from the tree and the states of the nodes in the tree are renewed.
4) Steps 2 and 3 are repeated until all PLPs have been traversed or all slices are allocated.

5.4 Simulations

The simulations in this chapter are performed using the same simulation model and video traffic as simulations in Chapters 3 and 4. Simulations are performed for an increasing number of services at different values of α (1 and 3). Performance is evaluated for D-StatMux, StatMux and MLWBOF for cases where there is number. EXP/PF has again been excluded because it has shown similar performance as MLWBOF in previous simulations. A data slice is formed by 2025 data cells, therefore, the total number of slices is $N_s = \left\lfloor \frac{(69 \times 27404)}{2025} \right\rfloor = 933$. The delay constraints for the services are given in Table 3.2 and services of the same category have the same delay constraint.

Simulation results obtained in chapters 3 and 4 assume that services are not delay constrained and all PLPs are allocated time slices successfully. In this chapter, delay constraints are imposed on the video services and are allocated time slices using the time slice allocation scheme described in Section 5.3. The time slice allocation scheme drops the BB frames from PLPs carrying services whose delay constraints and data rate requirements are not satisfied. Dropping of BB frames will directly affects the overall system throughput and total BB frame loss of the system. The effect of the time slice allocation to delay constrained services on throughput and BB frame loss is investigated in this chapter. Fairness performance is not included because time slice allocation has no influence on the way PLP schedulers allocate BB frames to PLPs. It must be noted that the time slice allocation is performed on the output PLPs of chapter 4. The system performance in the presence of time slice allocation is evaluated against when there is only PLP scheduling, and when there is both PLP and service scheduling.

According to Figure 5.5 including time slicing results in drop in throughput. This is because the time slice allocation drops the BB frames of PLPs whose delay and data rate requirements are not satisfied. When $\alpha = 1$, the throughput is low because the amount of traffic is low as only a subset of video layers is scheduled. There is a drop in throughput as a result of the BB

frames that are dropped consequent upon the fact that there are some PLPs that are not allocated time slices successfully. Not all PLPs in the same category are able to meet their delay and/or data rate requirements because there are not enough data slices before their deadlines. Therefore, the system throughput decreases. When $\alpha = 3$, the throughput is lower than when time slice allocation and service scheduling are not included in the system model, and when service scheduling is included, just like when $\alpha = 1$ as explained above. However, the results for when $\alpha = 3$ in Figure 5.5 is better in terms of throughput than when $\alpha = 1$. The reason for this is because with $\alpha = 3$, there is more data traffic in the system compared with when $\alpha = 1$.

The BB frame loss due to time slice allocation is shown in Figure 5.6. The schedulers show an increase in BB frame loss as α increases. The increase in BB frame loss is caused by the BB frames that are dropped when PLPs are unable to meet their data rate and delay constraints. The loss is lower at $\alpha = 1$ than $\alpha = 3$ because the traffic is fewer at the lower value of α and most services are able to meet both the delay and data rate requirements. When $\alpha = 1$ all the schedulers have a similar throughput and loss performance because all schedulers were able to satisfy data rate requirements of the PLPs without any BB frame loss when there are no delay constraints imposed. The time slice allocation scheme, therefore, operates on the same set of information. When $\alpha = 3$, BB frame loss performance is the same for all schedulers for up to 12 services because the time slice allocation scheme operates on the same set of information . As the services increase MLWBOF displays the best performance because it already prioritizes some services over other depending on their buffer states, and allocates more BB frames to them. And because the time slice allocation scheme gives priority to PLPs that require the most resources when the deadlines and weights are the same, MLWBOF is able to achieve a higher throughput and lower loss than other schedulers.

The simulation results show that time slice allocation compromises throughput and increases BB frame loss greatly. With time slice allocation the acceptable loss threshold of 5% is greatly exceeded. However, time slice allocation guarantees timely delivery of services which is essential when we have delay-sensitive services.

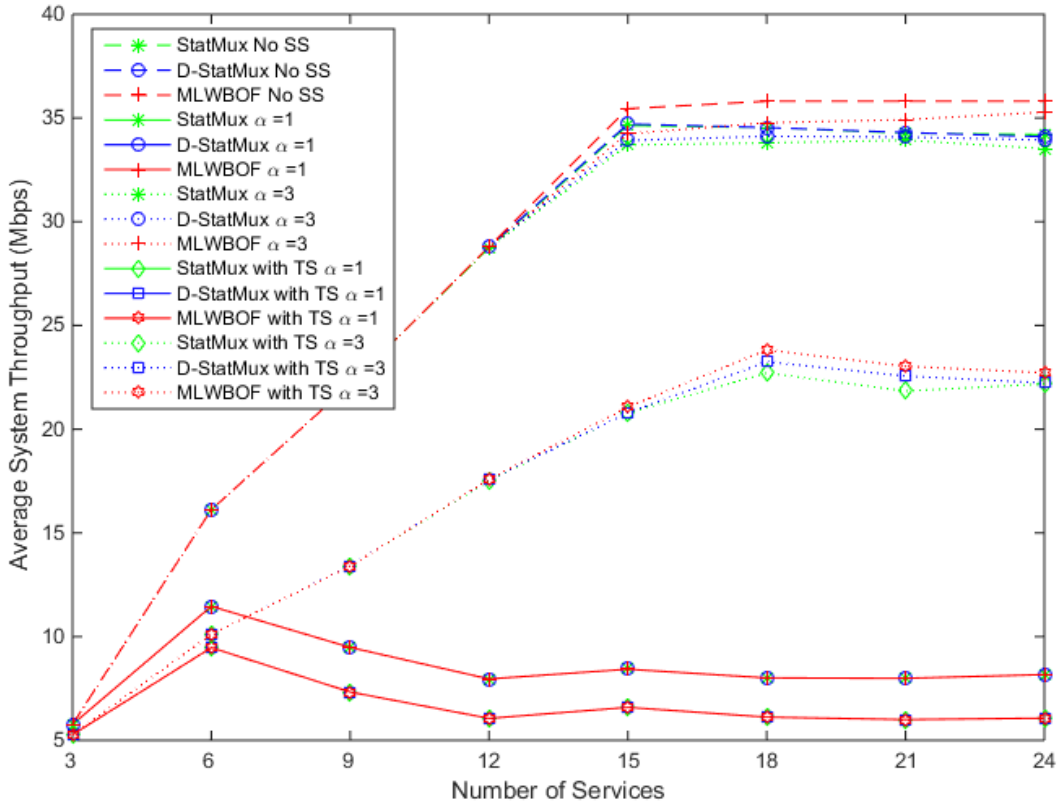


Figure 5.5 System Throughput with over-packing for Joint Service and PLP Scheduling with Time Slice Allocation

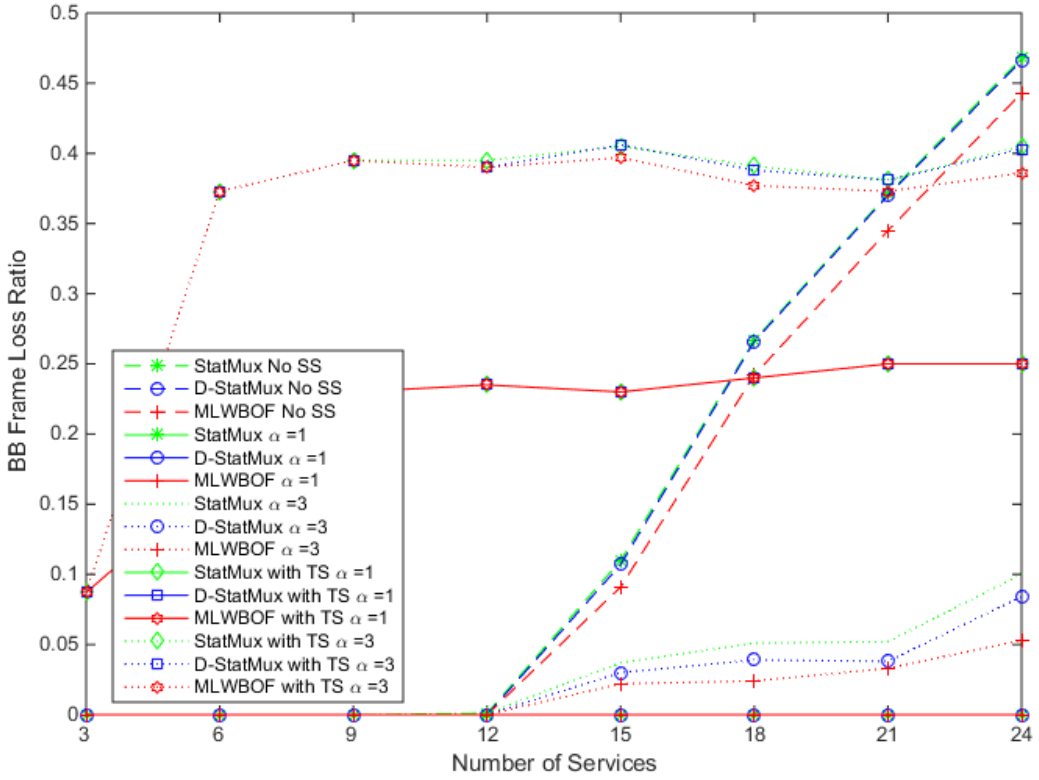


Figure 5.6 System BB Frame Loss due to Time Slice Allocation with over-packing for Joint Service and PLP Scheduling with Time Slice Allocation

5.5 Chapter Summary

In this chapter, time slice allocation is combined with service and PLP scheduling to ensure timely delivery of services. An existing time slice allocation scheme is employed and a detailed description of the scheme is presented. A system model for service and PLP scheduling with time slice allocation is described. Simulations are performed according to the system model and the same simulation model and performance measures as in chapters 3 and 4 are used. The results are analyzed and they show MLWBOF achieves a better performance in terms of throughput and total BB frame loss than the statistical multiplexers. Although time slice allocation guarantees timely delivery of services, it compromises throughput and BB frame loss. Therefore, a time slice allocation scheme that delivers services on time without greatly compromising throughput and BB frame loss performance of the system is necessary.

Chapter 6 Conclusions and Future Work

6.1 Conclusions

This dissertation aimed to improve the system performance in terms of throughput while considering QoS requirements and attaining fairness in DVB-T2. The main research question was formulated as follows: "How can SVC service be scheduled in DVB-T2 in order to make a good trade-off between throughput and fairness while considering QoS requirements?" The objectives of the study were to:

- Design and implement PLP scheduling schemes for BB frame allocation
- Design and implement a service scheduling scheme for bandwidth allocation to SVC services
- Design and implement a time slice allocation scheme in order to meet service delay constraints

Chapter 1 of the dissertation outlined to the background to the research problem by providing background information on DTTB and the latest DTTB standards, stating the research problem, specifying the research questions and laying out the research methodology. The structure of the dissertation was also outlined. Chapter 2 provided an overview of the DVB-T2 system and SVC as well as a review of the research literature relating to resource management in DVB-T2 and transmission SVC services over DVB-T2. The review was undertaken in order to identify the limitations of the existing resource management solutions in DVB-T2 which form the basis of this research. Chapter 3 proposed PLP scheduling schemes for BB frame allocation which aimed at achieving the first objective. The schemes are D-StatMux, MLWBOF and EXP/PF. D-StatMux was proposed as an improvement to the existing solutions while the other schedulers were adopted from LTE A system model was described and the performance of the PLP schedulers evaluated and the results were discussed. Chapter 4 presented a service scheduling scheme for allocation of bandwidth to SVC services as well as a system model. The service scheduler was adopted from LTE and exploits bit rate adaptability of SVC and allocates a subset of layers under bandwidth limitations. The chapter also dealt with the presentation and analysis of simulation results. Chapter 5 employed a time slice allocation scheme for DVB-T2 that exists in literature to evaluate how PLPs can be schedule in order to satisfy delay constraints. A system model was presented and simulation results were presented and interpreted. Chapter 6 provided a

conclusion which summarizes the work presented in the dissertation and made recommendations for future research.

The results obtained suggest that the proposed PLP schedulers perform better than existing scheduler in PLP Scheduling and joint service and PLP scheduling. MLWBOF and EXP/PF exhibit the best throughput and BB frame loss performances. The statistical multiplexers on the other hand have the best fairness performance and computational complexity. D-StatMux gave the best fairness performance. Service scheduling reduced BB frame loss at the expense of throughput and fairness especially for MLWBOF. Efficient service scheduling requires an optimal value of α that makes a good trade-off between throughput, fairness and BB frame loss needs to be identified. Introducing time slice allocation resulted in further decrease in throughput and increase in BB frame loss. The performance of schedulers is similar to the previous cases whereby the proposed schedulers give better throughput and loss performance than StatMux, and MLWBOF gives the best performance.

It can be concluded that the proposed PLP schedulers improve the DVB-T2 system performance in comparison with the existing solution. Although the statistical multiplexers have a lower complexity than MLWBOF and EXP/PF all the schedulers can easily be implemented in real-time. Service scheduling improves performance of the system because it enables an optimal trade-off between throughput, fairness and BB frame loss to be made. Time slice allocation deteriorates the performance of the system but it is essential when there are delay constraints to be satisfied. The results of the study were limited to only 3 video trace file samples from which 24 trace files were generated. The frame distribution of the generated trace files may have not been statistically independent hence the simulation results cannot be easily justified. Using trace files from 24 different video streams would produce better results. The results of this research were also limited to PLPs with the same physical layer configurations. The same acceptable BB frame loss threshold was used for all services which resulted in MLWBOF and EXP/PF giving the same performance. In spite of the limitations, the proposed PLP schedulers and service scheduler are recommended for use in DVB-T2.

6.2 Future Research

The following are suggested for future research:

a) Performing BB frame allocation and time slice allocation separately results in high overall BB frame loss and lower throughput. A scheduling scheme that considers both data rate requirement and delay constraint can be designed and implemented to reduce the BB frame loss and improve throughput. Also, when the number of services increases fairness is compromised greatly. A mechanism that makes a good trade-off between loss and throughput while maintaining fairness is necessary.

b) The scheduling problem can be approached as an optimization problem to maximize resource utilization under delay and data rate constraints. Another objective could be to minimize delay.

c) Since SVC videos are represented in multiple layers, the BB frames can be scheduled such that different layers are carried in different BB frames. This way BB frame dropping can be performed in such way that BB frame carrying upper layers are dropped first to ensure that highest possible quality is attained. For example, for a video with two quality layers the base layer packets can be delivered in odd BB frames while enhancement packets are delivered in even frames. The even BB frames can then be dropped before the odd BB frames.

d) The study can be extended to include a channel model and the effects of the scheduling schemes on the received video quality can be investigated.

References

[1] M. El-Hajjar and L. Hanzo, "A Survey of Digital Television Broadcast Transmission Techniques," *IEEE Communications Surveys & Tutorials,* vol. 15, pp. 1924-1949, 2013.

[2] ITU, *Handbook on Digital Terrestrial Television Broadcasting Networks and Systems Implementation*: ITU, 2016.

[3] ETSI, "TS 102 005 v1.4.1 (2010-03), Digital Video Broadcasting (DVB); Specifications for Use of Video and Audio Coding in DVB Services Delivered over IP Protocols ", ed: European Telecommunications Standard Institute, 2010.

[4] L. Vangelista, N. Benvenuto, S. Tomasin, C. Nokes, J. Stott, A. Filippi, *et al.*, "Key technologies for next-generation terrestrial digital television standard DVB-T2," *IEEE Communications Magazine,* vol. 47, pp. 146-153, 2009.

[5] ETSI, "Digital Video Broadcasting (DVB); Frame Structure Channel Coding and Modulation for Second Generation Digital Terrestrial Broadcasting System (DVB-T2)," ed: European Telecommunications Standards Institute, 2015.

[6] E. Tech, "3348 'Frequency and network planning aspects of DVB-T2'," *EBU Status Report,* vol. 4, 2014.

[7] R. Chernock and J. C. Whittaker, "Next-Generation Broadcast Television: ATSC 3.0 [Standards in a Nutshell]," *IEEE Signal Processing Magazine,* vol. 33, pp. 158-162, 2016.

[8] L. Fay, L. Michael, D. Gómez-Barquero, N. Ammar, and M. W. Caldwell, "An Overview of the ATSC 3.0 Physical Layer Specification," *IEEE Transactions on Broadcasting,* vol. 62, pp. 159-171, 2016.

[9] L. Fay, "ATSC 3.0 physical layer overview," in *2015 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting*, 2015, pp. 1-5.

[10] M. Takada and M. Saito, "Transmission System for ISDB-T," *Proceedings of the IEEE,* vol. 94, pp. 251-256, 2006.

[11] Z. Zi-Wei, Y. ZhiXing, P. Chang Yong, and Z. Yi-Sheng, "Synchronization and channel estimation for TDS-OFDM systems," in *2003 IEEE 58th Vehicular Technology Conference. VTC 2003-Fall (IEEE Cat. No.03CH37484)*, 2003, pp. 1229-1233 Vol.2.

[12] L. He, F. Yang, C. Zhang, and Z. Wang, "Synchronization for TDS-OFDM over multipath fading channels," *IEEE Transactions on Consumer Electronics*, vol. 56, pp. 2141-2147, 2010.

[13] J. Song and C. Zhang, "Technical Review on DTMB-Advanced (DTMB-A) Standard," in *2016 International Conference on Engineering and Telecommunication (EnT)*, 2016, pp. 128-133.

[14] S. Dardouri and R. Bouallegue, "Comparative Study of Downlink Packet Scheduling for LTE Networks," *Wireless Personal Communications*, vol. 82, pp. 1405-1418, June 01 2015.

[15] R. Basukala, H. A. M. Ramli, and K. Sandrasegaran, "Performance analysis of EXP/PF and M-LWDF in downlink 3GPP LTE system," in *2009 First Asian Himalayas International Conference on Internet*, 2009, pp. 1-5.

[16] F. Afroz, K. Sandrasegaran, and P. Ghosal, "Performance analysis of PF, M-LWDF and EXP/PF packet scheduling algorithms in 3GPP LTE downlink," in *2014 Australasian Telecommunication Networks and Applications Conference (ATNAC)*, 2014, pp. 87-92.

[17] F. Capozzi, G. Piro, L. A. Grieco, G. Boggia, and P. Camarda, "Downlink Packet Scheduling in LTE Cellular Networks: Key Design Issues and a Survey," *IEEE Communications Surveys & Tutorials*, vol. 15, pp. 678-700, 2013.

[18] L. Kondrad, I. Bouazizi, V. K. M. Vadakital, M. M. Hannuksela, and M. Gabbouj, "Cross-layer optimized transmission of H.264/SVC streams over DVB-T2 broadcast system," in *2009 IEEE International Symposium on Broadband Multimedia Systems and Broadcasting*, 2009, pp. 1-5.

[19] I. Eizmendi, M. Velez, D. Gómez-Barquero, J. Morgade, V. Baena-Lecuyer, M. Slimani, *et al.*, "DVB-T2: The Second Generation of Terrestrial Digital Video Broadcasting System," *IEEE Transactions on Broadcasting*, vol. 60, pp. 258-271, 2014.

[20] ETSI, "Digital Video Broadcasting (DVB); Implementation Guidelines for a Second Generation Digital Terrestrial Broadcasting System (DVB-T2) ", ed: European Telecommunications Standards Institute, 2012.

[21] M. Rezaei, I. Bouazizi, and M. Gabbouj, "Statistical Time-Frequency Multiplexing of HD Video Traffic in DVB-T2," *International Journal of Digital Multimedia Broadcasting*, vol. 2009, p. 12, 2009.

[22] J. P. Bou, "Physical Layer Statistical Multiplexing for the Second Generation Digital Terrestrial TV Standard DVB-T2."

[23] X. D. Yang, Y. H. Song, T. J. Owens, J. Cosmas, and T. Itagaki, "Performance analysis of time slicing in DVB-H," in *Mobile Future, 2004 and the Symposium on Trends in Communications. SympoTIC '04. Joint IST Workshop on*, 2004, pp. 183-186.

[24] F. Wang, Z. Wang, and S. Chen, "Efficient and reliable slice allocation for multi-services in DVB-T2 networks," *IET Communications,* vol. 11, pp. 837-845, 2017.

[25] H. Schwarz, D. Marpe, and T. Wiegand, "Overview of the Scalable Video Coding Extension of the H.264/AVC Standard," *IEEE Transactions on Circuits and Systems for Video Technology,* vol. 17, pp. 1103-1120, 2007.

[26] I. Unanue, I. Urteaga, R. Husemann, J. Del Ser, V. Roesler, A. Rodriguez, *et al.*, "A Tutorial on H.264/SVC Scalable Video Coding and its Tradeoff between Quality, Coding Efficiency and Performance," in *Recent Advances in Video Coding*, ed: InTech, 2011.

[27] M. Wien, H. Schwarz, and T. Oelbaum, "Performance Analysis of SVC," *IEEE Transactions on Circuits and Systems for Video Technology,* vol. 17, pp. 1194-1203, 2007.

[28] C. Hellge, T. Wiegand, E. G. Torre, D. Gomez-Barquero, and T. Schierl, "Efficient HDTV and 3DTV Services over DVB-T2 using multiple PLPs with layered media," *IEEE Communications Magazine,* vol. 51, pp. 76-82, 2013.

[29] M. Tormos, C. Tanougast, P. Bretillon, J. F. Travers, P. Kasser, and A. Dandache, "Performance evaluation of SVC coding using MPLP-DVB-T2 for mobile and fixed reception," in *IEEE international Symposium on Broadband Multimedia Systems and Broadcasting*, 2012, pp. 1-6.

[30] A. Ksentini and Y. Hadjadj-Aoul, "On Associating SVC and DVB-T2 for Mobile Television Broadcast," in *2011 IEEE Global Telecommunications Conference - GLOBECOM 2011*, 2011, pp. 1-5.

[31] ETSI, "Digital Video Broadcasting (DVB); Measurement Guidelines for DVB Systems," ed: European Telecommunications Standards Institute, 2014.

[32] P. Seeling and M. Reisslein, "Video Transport Evaluation With H.264 Video Traces," *IEEE Communications Surveys & Tutorials,* vol. 14, pp. 1142-1165, 2012.

[33] R. Gupta, A. Pulipaka, P. Seeling, L. J. Karam, and M. Reisslein, "H.264 Coarse Grain Scalable (CGS) and Medium Grain Scalable (MGS) Encoded Video: A Trace Based Traffic and Quality Evaluation," *IEEE Transactions on Broadcasting*, vol. 58, pp. 428-439, 2012.

[34] *Video Trace Library*. Available: http://traces.aes.asu.edu

[35] R. Jain, D.-M. Chiu, and W. R. Hawe, *A quantitative measure of fairness and discrimination for resource allocation in shared computer system* vol. 38: Eastern Research Laboratory, Digital Equipment Corporation Hudson, MA, 1984.

[36] A. Orsino, P. Scopelliti, and M. Condoluci, "A Multi-Criteria Approach for Multicast Resource Allocation over LTE and Beyond Cellular Systems," in *European Wireless 2016; 22th European Wireless Conference*, 2016, pp. 1-6.

[37] C. N. Lee and H. T. Lai, "Pricing based resource allocation scheme for video multicast service in LTE networks," in *2016 Asia-Pacific Signal and Information Processing Association Annual Summit and Conference (APSIPA)*, 2016, pp. 1-5.

[38] M. Condoluci, G. Araniti, A. Molinaro, and A. Iera, "Multicast Resource Allocation Enhanced by Channel State Feedbacks for Multiple Scalable Video Coding Streams in LTE Networks," *IEEE Transactions on Vehicular Technology*, vol. 65, pp. 2907-2921, 2016.

[39] P. Chau, Y. Lee, T. D. Bui, J. Shin, and J. Jeong, "An efficient resource allocation scheme for scalable video multicast in LTE-advanced networks," presented at the Proceedings of the 11th International Conference on Ubiquitous Information Management and Communication, Beppu, Japan, 2017.