QoS Aware Packet Scheduling in the Downlink of LTE-Advanced Networks

Rehana Kausar

Submitted for the Degree of Doctor of Philosophy

School of Electronic Engineering & Computer Science
Queen Mary University of London

January 2013

To my parents

Abstract

This thesis considers QoS aware packet scheduling in the downlink of LTE-Advanced networks. The outcome of this research is improved QoS performance for real-time services while maintaining the overall system-level performance.

A QoS aware scheduling architecture is proposed that extends the traditional MAC layer scheduling into a three-stage cross-layer architecture that takes information from the Physical, MAC and application layers. The system includes:

- two service-specific queue-sorting algorithms for both real-time and non-real time service in the traffic differentiating stage to prioritise users;
- an adaptive scheduling algorithm in the time-domain scheduling stage that incorporates machine learning to achieve adaptive resource allocation; and
- a frequency domain scheduling stage using a modified proportional fairness algorithm for resource allocation.

The first two stages are completely novel, as is the combination with the frequency domain scheduling which uses modified proportional fairness algorithm.

The performance of the proposed algorithms is evaluated in a system-level simulator under different network scenarios. Simulation results show that the approach can (i) reduce the average delay of the real-time service and fulfil the minimum throughput requirements of the non-real time service and (ii) achieve a good trade-off between user-level and system-level performance.

Acknowledgements

All praises and thankfulness is for Allah the Almighty who gave me the courage, determination and steadfastness to complete this thesis. I would like to present my humble thanks to my great religious leader for his precious prayers and also for his financial support during my study, may Allah give him the best reward.

I would like to take this opportunity to express my sincere gratitude to my supervisor, Dr Yue Chen, for her invalueable academic support. Her supervision, apart from academics, groomed me logically and always guided me towards positive and progressive thinking. Her advice has always been the greatest source of motivation for me. Also I would like to thank Dr Michael Chai for his continous support and encouragement at every stage of my study. I would like to thank Prof Laurie Cuthbert, Dr John Schormans, Dr Silvano P V Barros (late) and Dr Vindya Wijeratne for their help and support.

I wish to extend my gratitude to all the support staff of the department, who have made the technical part of my work possible. Many thanks to my colleagues and friends who have encouraged and advised me and made my time here memorable.

Finally, I would like to thank my husband Mansoor Ahmed and my family for their love, continous prayers and enourmous support. They always be my strongest power to work hard and get through hard times. With all my love and gratitude, I want to dedicate this thesis to my father Jamil Ahmad Tahir (late) who had a great wish to see me as doctor and to my mother Amtul Hai for her continuous prayers.

Declaration

I hereby declare that the work presented in this thesis is solely my work and that to the best of my knowledge the work is original except where indicated by reference to the respective authors.

Rehana Kausar

Date:

Table of Contents

List of Figures

List of Tables

List of Abbreviations

1G	First Generation
2G	Second Generation
3G	Third Generation
4G	Fourth Generation
ACK	Acknowledgment
ATDSA	Adaptive Time Domain Scheduling Algorithm
ARP	Allocation and Retention Policy
BCCH	Broadcast Control Channel
BE	Best Effort
BS	Base Station
CLPSA	Cross Layer Packet Scheduling Architecture
CQI	Channel Quality Information
DB	Delay Budget
DL	Downlink
DL-SCH	Downlink Shared Channel
EDGE	Enhanced Data Rates for GSM Evolution
eNB	Evolved NodeB
FD	Frequency Domain
FDMA	Frequency Division Multiple Access
FD-PS	Frequency Domain Packet Scheduling
GBR	Guaranteed Bit Rate
GSM	Global System for Mobile Communications
GPRS	General Packet Radio Service
HARQ	Hybrid Automatic Repeat Request
HOL	Head of Line

HSCSD	High Speed Circuit Switch Data
IMAP	Internet Message Access Protocol
IP	Internet Protocol
IMS	IP Multimedia Subsystem
ITU	International Telecommunication Unit
LAN	Local Area Network
LOS	Line-of-Sight
LTE	Long Term Evolution
MAX C/I	Maximum Carrier to Interference
MBR	Maximum Bit Rate
MIX	Mix Traffic Packet Scheduling
M-LWDF	Modified-Longest Waited Delay First
MMS	Multimedia Messaging Service
MU	Multiuser
NAK	Not Acknowledged
NLOS	Non Line-of-Sight
NRT	Non Real time
OFDMA	Orthogonal Frequency Domain Multiple Access
PCFICH	Physical Control Format Indicator Channel
PDA	Personal Digital Assistants
PDCCH	Physical Downlink Control Channel
PDSCCH	Physical Downlink Shared Channel
PF	Proportional Fair
PDR	Packet Drop Rate
POP	Post Office Protocol
PRB	Physical Resource Block
PS	Packet Scheduling

QCI	QoS Class Identifier
QMA	Queue Management Algorithm
QoS	Quality of Service
QSI	Queue State Information
RR	Round Robin
RT	Real Time
RRM	Radio Resource Management
RX	Receiver
SC-FDMA	Single Carrier FDMA
SMS	Short Messaging Service
SNR	Signal to Noise Ratio
SSSA	Service Specific Queue-Sorting Algorithms
SWBS	Sum Waiting Time Based Scheduling
TCP	Transport Control Protocol
TD	Time Domain
TTI	Transmission Time Interval
TTI	Transmitter
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunication Services
VoIP	Voice over IP
WCDMA	Wideband Code-Division Multiple Access

Chapter 1 Introduction

This chapter introduces the work presented in this thesis by briefly describing the background knowledge and motivation of the work. The importance of the work is described in the research scope section followed by the novel contributions of this thesis.

1.1 Background and Motivation

Long Term Evolution-Advanced (LTE-A) networks are envisioned to support wide range of multimedia applications such as voice telephony, internet browsing, interactive gaming, video messaging, email, etc. [3GP09]. These applications demand different Quality of Service (QoS) requirements, such as average packet delay, average Packet Drop Rate (PDR), and minimum throughput requirements [TCT10]. To fulfil diverse QoS requirements is more challenging in wireless networks as compared to the wired networks. This is due to the limited radio resources, the time-varying channel conditions and resource contention among multiple users.

Packet scheduling deals with radio resource allocation and is directly related to the QoS provision to users demanding different services. Three classic packet scheduling algorithms are Round Robin (RR), Maximum Carrier to Interference (MAX C/I) and Proportional Fairness (PF). These algorithms consider system-level performance such as fairness and system spectral efficiency, and do not consider the QoS requirements of different services from individual users. The current state of the art packet scheduling algorithms mainly take QoS provision into consideration in order to provide better service experience for each individual user. However, the performance of QoS aware packet scheduling algorithms at both user-level and system-level has not been addressed properly.

The joint consideration of QoS provision at individual user-level and the overall system-level performance is very crucial for an effective resource allocation. In this thesis a novel QoS aware packet scheduling architecture is proposed which aims at improving the QoS provision to different services while achieving a good trade-off between user-level and system-level performance. The user-level performance is investigated by the QoS provision to different services and for the system-level performance, system throughput, throughput fairness among all users and average PDR fairness among all Real Time (RT) users are considered.

1.2 **Research Scope**

This thesis describes research into the QoS aware scheduling in the Downlink (DL)of Orthogonal Frequency Division Multiple Access (OFDMA) based LTE-A networks, by addressing both user-level and system-level packet scheduling performance.

A cross-layer packet scheduling architecture is proposed and designed, which takes information on service types, QoS requirements, queue states and channel states from different protocol layers to take scheduling decisions.

At the user-level, a *Traffic Differentiator* stage is applied to segregate packet queues from all active users into different service queues based on their service types. In each service queue, users are prioritised according to their QoS requirement and wireless channel conditions by implementing novel service-specific queue-sorting algorithms (SSSA).

At the system-level, fairness among users is one of the key attributes to be considered. In multi-service networks such as LTE-A, it is important to allocate as fair share of available radio resource as possible among all users, to avoid any service starvation. This thesis proposes a novel Adaptive Time Domain Scheduling Algorithm (ATDSA) in the *TD*

Scheduler stage, which aims at allocating just enough radio resource to real-time and non-real time services, and assigning the remaining available resource to the background service. This is achieved by integrating Hebbian learning process and K-mean clustering algorithm in ATDSA. Hebbian learning is applied to improve throughput fairness among all users and K-mean clustering is applied to improve the average PDR fairness among users requiring real-time service.

The *FD Scheduler* stage uses modified PF algorithm for resource allocation to exploit Frequency Domain (FD) Multi User (MU) diversity.

1.3 **Contributions**

The work presented in this thesis is novel. The main contributions are:

A new cross-layer design QoS aware packet scheduling architecture for the DL transmission of LTE-A networks, which incorporates different queue-sorting algorithm for each service and integrates machine learning in the Time Domain (TD), to take adaptive scheduling decisions. The feedbacks on Channel State Information (CSI) and information on parameters such as QoS measurements and Queue State Information (QSI) are used at different stages of the proposed architecture to make scheduling decisions adaptive to the service demands and changing network scenarios.

- A service-specific queue sorting algorithm is proposed for real-time voice service to prioritise Real Time (RT) users based on their QSI including, Head of Line (HOL) packet delay, delay budget and queue length, and CSI. This algorithm leads to reduced, average packet delay, delay variation and average packet drop rate of real-time service.

- A service specific queue sorting algorithm is proposed for non-real time streaming video service to prioritise Non Real Time (NRT) users based on the minimum throughput requirements of this service, QSI including HOL packet delay and delay budget and CSI. This algorithm meets the QoS requirements of streaming video service better compared with conventional packet scheduling algorithms.

- An adaptive time domain scheduling algorithm is proposed, which integrates Hebbian learning process to adaptively allocate the available radio resource to all services based on the average PDR information of real-time voice and non-real time streaming video services. It allocates just enough radio resource to real-time and non-real time services to meet their required QoS, and assigns the rest to the background service. This algorithm with learning capability, leads to improved QoS provision to real-time voice service and non-real time streaming video service. At the same time, it improves the fairness at system-level by preventing background service from starvation.

- K-mean clustering algorithm is integrated into the time domain scheduling to improve system-level performance in terms of average PDR fairness among RT users. It leads to a significant reduction in average PDR variation of RT users, especially in highly loaded network scenarios.

1.4 Author’s Publication

[Re-1] Rehana Kausar, Yue Chen, Kok. Keong. Chai, Laurie Cuthbert, John Schormans, “ QoS aware Mixed Traffic Packet Scheduling in OFDMA-Based LTE-Advanced Networks”, Fourth International Conference on Mobile Ubiquitous Computing Systems, Services and Technologies 2010 (UBICOMM 2010), Florence, Italy, October 25-30, 2010, [Best paper award].

[Re-2] Rehana Kausar, Yue Chen, Kok. Keong. Chai, “Service Specific Queue Sorting and Scheduling Algorithm for OFDMA-Based LTE-Advanced Networks”, Sixth International Conference on Broadband and Wireless Computing, Communication and Applications (BWCCA), Barcelona, Spain, October 26-28, 2011.

[Re-3] Rehana Kausar, Yue Chen, Kok. Keong. Chai, “Adaptive Time Domain Scheduling Algorithm for OFDMA Based LTE-Advanced Networks”, IEEE Seventh International Conference on Wireless and Mobile Computing (WiMob), Networking and Communications, 10 - 12 October 2011.

[Re-4] Rehana Kausar, Yue Chen, Kok. Keong. Chai, “ QoS Aware Packet Scheduling with Adaptive Resource Allocation for OFDMA Based LTE-Advanced Networks”, IET International Conference on Communication Technology and Application (ICCTA), Beijing, China, 14-16 Oct 2011.

[Re-5] Rehana Kausar, Yue Chen, Kok. Keong. Chai, John Schormans, “ QoS Aware Packet Scheduling Framework with Service Specific Queue Sorting and Adaptive Time Domain Scheduling Algorithms for LTE-Advanced Downlink”, International Journal on Advances in Networks and Services, ISSN 1942-2644, Volume 4, numbers 3 & 4, pp. 244-256, 2011.

[Re-6] Rehana Kausar, Yue Chen, Kok. Keong. Chai, John Schormans, " QoS Aware Intelligent Scheduling Architecture for LTE-Advanced Networks", The Research to Industry Conference Henry Ford College, Loughborough (R2i)– 19th June 2012, an invited poster and oral presentation.

[Re-7] Rehana Kausar, Yue Chen, Kok. Keong. Chai, "An Intelligent Scheduling Architecture for Mixed Traffic in LTE-Advanced", 23rd IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC) 2012. Sydney Australia, 9-12 Sep. 2012.

[Re-8] Rehana Kausar, Yue Chen, Kok. Keong. Chai, ""Broadband Wireless Access Networks for 4G: Theory, Applications, and Experimentation", IGI Global (formerly Idea Group Inc.), contribution of one chapter accepted.

1.5 Thesis Organisation

The remaining thesis is structured as follows.

Chapter two presents background information on access technologies in cellular networks, the principle of packet scheduling and the QoS architecture in LTE-A networks. This chapter also describes classic packet scheduling algorithms and state-of-the-art on QoS aware packet scheduling. In addition it gives an overview of incorporating machine learning processes in scheduling with some learning algorithms.

Chapter three describes the research contributions of this thesis in detail. It describes the cross-layer concept used in this research and the functionality of each stage of the proposed packet scheduling architecture. The principle of all novel algorithms proposed at different stages of the scheduling architecture such as algorithms in SSSA and ATDSA, are explained in detail. At the end system-level performance evaluation indicators are given, which are used to evaluate the performance of the proposed algorithms.

Chapter four describes the setup of the system-level simulation platform, used in this thesis for performance evaluation. It includes the overall simulation structure, simulation parameters and detailed functionality of the main simulation modules. It also describes the wireless channel model used for this research. The important processes of validation and verification, essential when using simulation-based research, are also described in this chapter.

Chapter five presents simulation results from all proposed algorithms and their analysis, to evaluate the proposed scheduling architecture. It presents simulation results of SSSA, ATDSA, separately and jointly. The performance of the whole overall architecture is analysed under different network scenarios. The simulation results are compared against the state-of-the-art QoS aware Sum Waiting Time Based Scheduling (SWBS) and Mix Traffic Packet Scheduling for UTRAN Long Term Evolution Downlink (abbreviated as MIX hereafter) scheduling algorithms.

Chapter six concludes the work in this thesis. It presents the significant results and discusses the potential future work.

Chapter 2 Research Study

This chapter describes in detail the fundamental technologies used in resource allocation in different generations of mobile networks. The conventional and QoS-aware state-of-the-art packet scheduling algorithms are investigated including machine learning based scheduling in wireless networks. It also presents a study of service classes and service requirements according to 3GPP specifications.

2.1 Access Technologies in Wireless Mobile Networks

Radio resource allocation schemes are closely related to the radio access technologies used in mobile networks. In the last few decades, various radio access technologies have been developed to enhance the resource allocation performance in mobile communication networks. Figure 2-1 illustrates radio access technologies used in different generations of mobile networks [GPKM08].

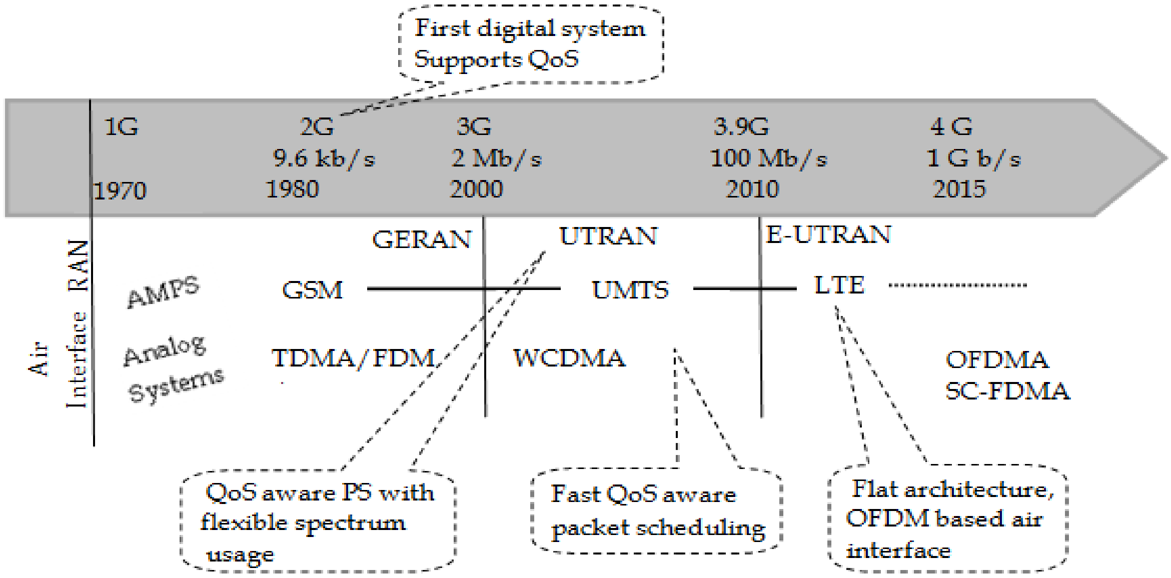
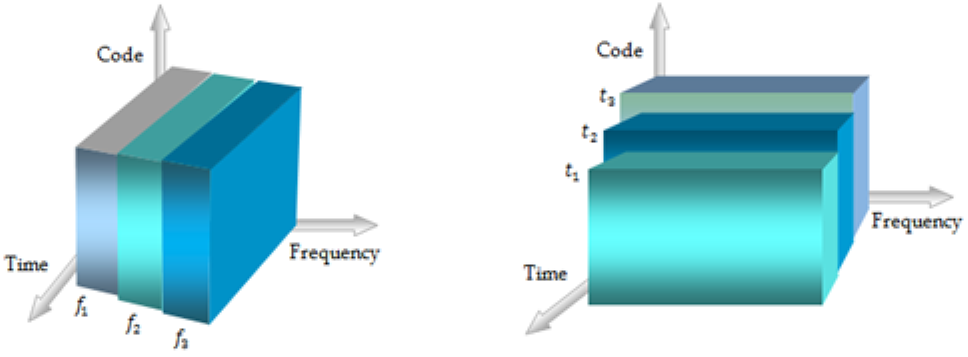


Figure 2-1 Evolution of access technologies in mobile cellular networks

The first generation (1G) radio communication systems like Advanced Mobile Phone System (AMPS) or Nordic Mobile Telephone (NMT) were launched in 1970s, which use simple analogue communication technology to provide voice call services [Yang05]. The second generation (2G), Global System of Mobile Communications (GSM) commercially started in 1980s and it uses digital communication technique. GSM became the most commercially successful 2G system as it was the first fully specified system with international compatibility and transparency [KALNN05]. Until GSM, mobile networks had a circuit switched core network. However in 1990s, a packet switched core network was added on the top of the traditional circuit switched GSM network under the name General Packet Radio Service (GPRS). It started to provide basic packet based services to the mobile users such as Internet over the Wireless Application Protocol (WAP). However in the first version of GPRS, QoS was supported only at the core network level as the GSM radio interface was designed for circuit switched connections [KALNN05]. The development of the third Generation (3G) systems was steered by the increasing demand of higher data rates. The international standardisation body 3GPP (third Generation Partnership Project) started specifications in 1998 of the Universal Mobile Telecommunication System (UMTS) with a new air interface Wideband Code Division Multiple Access (WCDMA), for 3G networks [Kumar09]. WCDMA allows a very flexible use of the available spectrum because of advanced power control and Link Adaptation (LA) techniques and Universal Terrestrial Radio Access Network (UTRAN). The flexibility of radio air interface makes QoS aware Radio Resource Management (RRM) very crucial. High Speed Downlink Packet Access (HSDPA) in an enhancement brought to UTRAN. HSDPA, among other technologies, brings Packet Scheduler (PS) and LA closer to the air interface in the Base Station (BS), which allows a faster adaptation to the channel conditions hence more flexibility and data rates [HT07].The fourth Generation (4G) systems formalise the convergence between mobile networks and wireless Local Area Network (LAN) systems into "broadband wireless access" [Adachi02]. 3GPP has defined the standardisation for all Internet Protocol (IP) mobile network systems Long Term Evolution (LTE), which applies the new RAN Evolved-UTRAN (E-UTRAN). The E-UTRAN uses a simplified core network and RAN architecture in order to reduce latency of packet based services. The new RAN is composed of only one node, the evolved Node B (eNodeB/e-NB), which carries all RRM functionalities. LTE will finally evolve into LTE-A networks.

The 2G GSM system applies both Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) access technologies to allocate radio resource.

Figure 2-2 shows the principle of FDMA and TDMA technologies.



(a) FDMA

(b) TDMA

Figure 2-2 FDMA and TDMA technology

In FDMA, signals for different users are transmitted on different frequency bands at the same time. In TDMA, signals for different users are transmitted in the same frequency band at different times [JWY05].

GSM network uses a combination of both FDMA and TDMA as air interface. The whole spectrum is divided into fourteen bands with different frequencies. Each frequency band is divided into eight time slots. Different users are allocated with different channels which occupy different time slots on the same or different frequency bands.

In the 3rd Generation (3G) mobile cellular networks, Code Division Multiple Access (CDMA) is adopted as the radio access technology [AG11]. Various standards are developed for 3G systems which are used in different countries, such as WCDMA and Time Domain-Synchronous Code Division Multiple Access (TD-SCDMA) for UMTS, and Multi-Carrier CDMA (MC-CDMA) for CDMA2000 [JWY05] [OLDSMITH05]. The general principle of CDMA is shown in Figure 2-3.

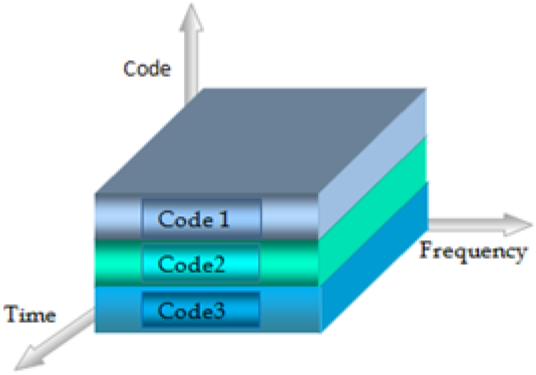


Figure 2-3 CDMA multiple access technology in 3G networks

In CDMA, signals for different users are transmitted in the same frequency band at the same time but using different codes. As all signals are transmitted at same time and on the same band so the capacity is mainly constrained by the interference between signals of different users.

2.2 OFDMA in Long Term Evolution (LTE) Networks

LTE networks use OFDMA as the DL transmission technology and Single Carrier Frequency Division Multiple Access (SC-FDMA) in the Uplink (UL) transmission technology [HT09]. These new technologies enable LTE networks to support a wide range of applications and to meet the requirements set by 3GPP in terms of peak data rate, which is 100 Mbps for the DL and 50 Mbps for UL and 1.5bps/Hz spectrum efficiency in a 20 MHz bandwidth. In some countries, like the US and the UK, LTE networks have been commercialised. For further development, 3GPP has started defining LTE-A in Release 10, which is expected to achieve peak data rate up to 3 Gbps for the DL and 1.5 Gbps for the UL and 30 bps/Hz spectrum efficiency in a 20 MHz bandwidth [KPKRHM08].

As this thesis only considers the DL transmission in LTE-A networks, therefore the DL radio access technology OFDMA is describes in detail.

OFDMA is a multiple-access version of Orthogonal Frequency Division Multiplexing (OFDM) scheme [Rappaport02]. The principle of OFDM is to use narrow, mutually orthogonal sub-carriers to carry data, and OFDMA is achieved by assigning sub-channels to carry data from different users. The OFDMA system assigns a subset of sub-carriers (termed as OFDMA channel) to individual users. Each traffic channel is assigned exclusively to one user at any time [3GPP09], as shown in Figure 2-3. In an OFDMA system, users are not overlapped in the frequency domain at any given time; however, the frequency band allocated to a particular user may change over the time. Figure 2-4 shows the principle of radio resource allocation in OFDMA technology.

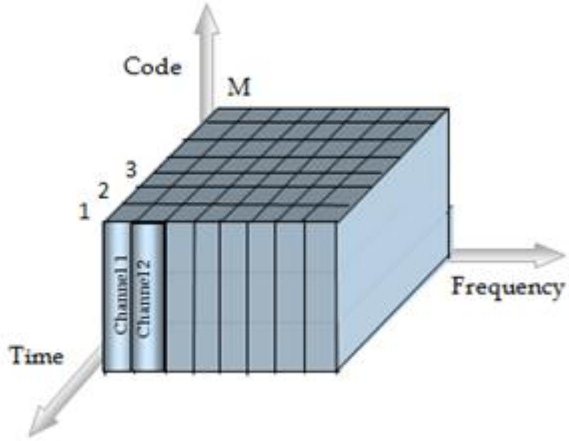


Figure 2-4 OFDMA technology

OFDMA is suitable for high data rate transmission in wideband wireless systems due to its spectral efficiency and good immunity to multipath fading [AM08]. Moreover this technology also provides a possible further enhancement to the system by enabling opportunistic scheduling in the frequency domain. To frequency multiplex users in LTE, sub-carriers are divided into sub-channels and each sub-channel or Physical Resource Block (PRB) is composed of several neighbouring sub-carriers. By grouping the sub-carriers into sub-channels or PRBs, the amount of control signalling and complexity of scheduling can be reduced considerably [DPSB08]. Frequency domain packet scheduling which is used in LTE is a powerful technique for improving the system capacity.

An important aspect of using OFDMA in LTE-A is that allocation is not done on individual sub-carrier basis but is based on PRB. Each PRB consists of 12 sub-carriers and is equivalent to a minimum bandwidth allocation of 180 kHz, where the respective allocation resolution in the time domain is 1ms. The downlink transmission allocation thus means filling the resource pool with 180 kHz blocks at 1 ms resolution as shown in Figure 2-5 [3GPP11].

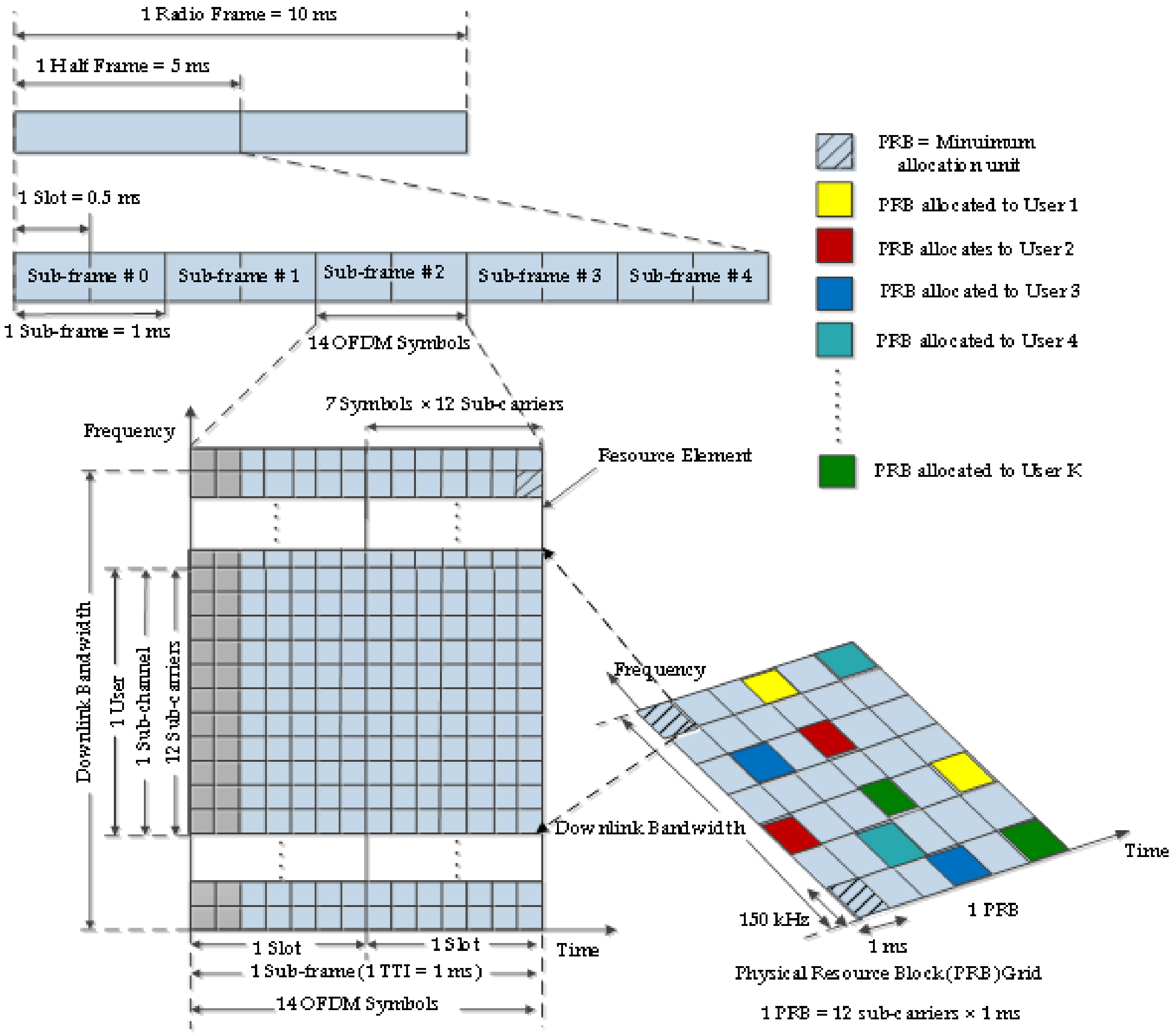


Figure 2-5 The OFDMA technology in LTE/LTE-A

In LTE/LTE-Advanced, a PHY frame consists of ten sub-frames in the time domain [3GPP12], where each sub-frame (i.e., one Transmission Time Interval, TTI) has two slots to carry 14 OFDM symbols, as shown in Figure 2-5. In frequency domain, a PHY channel is divided into a number of sub-channels, where each sub-channel is a group of sub-carriers. This element of allocating resources in the FD is often referred to as FD scheduling or FD diversity [Rappaport02]. The minimum allocation unit in LTE/LTE-Advanced is 1 PRB, which is allocated to users. The LTE/LTE-Advanced scheduling algorithm schedules users' data symbols onto a PRB having a resolution of one sub-frame (in time 1 ms) and one sub-channel (in frequency 180kHz). Different bandwidth spectrums use different number of PRBs, i.e., a wider bandwidth spectrum includes more PRBs. For example, in the lowest bandwidth of 1-4 MHz spectrum, a single channel consists of 6 PRBs and in the highest bandwidth of 20 MHz, a single channel consists of 100 PRBs [3GPP11a].

2.3 Packet Scheduling

In a RAN, RRM is the set of components that helps achieving the required QoS while efficiently using the available radio resources. Packet scheduling is one of the main RRM functionalities and it is responsible for the selection of users and transmission of their packets such that the available radio resource is efficiently utilised and the users' QoS requirements are satisfied.

Scheduling process deals with assigning the portions of available spectrum shared among users, by following specific policies [TCT10]. The policy refers to the decision process used to choose which users should be allocated radio resources in the given TTI and which users should be delayed to the next TTI [Proacis01] [Jayakumari10] in order to provide the required QoS, fairness, system spectral efficiency and service priorities. Figure 2-6 shows the concept of basic scheduling process.

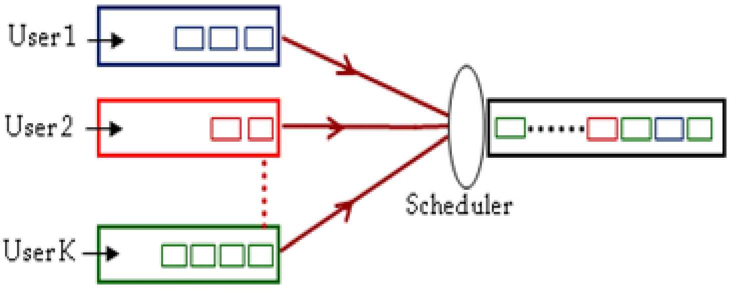


Figure 2-6 Basic model of a packet scheduler

The principle of a scheduler is to schedule users demanding different services, according to the pre-defined packet scheduling algorithms, to enhance QoS provision to users and the system-level performance [ZP02]. These users demand different QoS requirements e.g. delay and throughput. The scheduler takes into account of these requirements and schedules users accordingly. Packets from the user queues are transmitted based on scheduler decisions. Packet Scheduling is thus a method of network bandwidth management that can monitor the importance of data packets and depending upon the priority of the packet, give it higher or lower priority.

In [KSA08][HT09], a dynamic packet scheduler is defined as the main entity to take scheduling decisions dynamically to ensure high spectral efficiency while providing required QoS.

Different performance indicators have been used for evaluating performance of packet schedulers in various research works. Main three are summarised as below.

- **QoS provision to different services:** A QoS framework is a fundamental component of the next generation broadband networks for satisfactory service delivery of evolving Internet applications to the end users, and managing the network resources. Today's popular mobile Internet applications, such as voice, gaming, streaming, and social networking services, have diverse traffic characteristics and, consequently, demand different QoS requirements. The data traffic associated with these services must be delivered to the end users at specific data rates and/or within specific delay, packet loss and delay variation bounds. These requirements can collectively be termed as QoS. A rather flexible QoS framework is highly desirable to be future-proof to deliver the incumbent as well as emerging mobile Internet applications.

- **System spectral efficiency:** System spectral efficiency is typically measured in bit/s/Hz (bit per second per Hz), bit/s/Hz/cell (bit per second per Hz per cell) or bit/s/Hz/site (bit per second per Hz per site). It is a measure of the quantity of users or services that can be simultaneously supported by limited radio frequency bandwidth in a defined geographic area. It may for example be defined as the maximum throughput, summed over all users in the system, divided by the channel bandwidth.

- **Fairness among users:** Fairness measures or metrics are used in network engineering to determine whether users or applications are receiving a fair share of system resources. In packet radio wireless networks, the fairly shared spectrum efficiency (FSSE) can be used as a combined measure of fairness and system spectrum efficiency. The system spectral efficiency (as described before) is the aggregate throughput in the network divided by the utilized radio bandwidth in hertz. The FSSE is the portion of the system spectral efficiency that is shared equally among all active users. In case of scheduling starvation, the FSSE would be zero during certain time intervals. In case of equally shared resources, the FSSE would be equal to the system spectrum efficiency. Another fairness indicator is average PDR fairness, which is achieved by reducing the variation in average PDR values of users.

In this thesis, the most commonly used performance metrics are adopted to validate the effectiveness of the proposed scheduling architecture. They include average packet delay, average PDR, throughput for different types of services, overall system throughput and fairness.

2.4 Scheduling in LTE-A Networks

In LTE-A networks, all scheduling decisions whether it is UL or DL, are taken by eNB, which is commonly known as Base Station (BS) in previous mobile generations. One of the basic principles of the LTE/LTE-A radio access is shared-channel transmission which means time-frequency resources are dynamically shared between users [DPSB08]. The scheduler is a part of Medium Access Control (MAC) layer and controls the assignment of uplink and downlink resources. The basic operation of the scheduler is so-called dynamic scheduling, where eNB makes scheduling decisions and sends the scheduling information to the selected set of terminals. The downlink scheduler is responsible for dynamically controlling the terminals to transmit to; the set of resource blocks upon which the terminal's DL Shared Channel (DL-SCH) should be transmitted. The Physical Downlink Shared Channel (PDSCH) carries the user data rate and the Physical Downlink Control Channel (PDCCH) informs the device which resource blocks are allocated to it [DPSB08], dynamically with 1 ms granularity. For the channel-dependent scheduling, the mobile terminals transmit channel status reports reflecting the instantaneous channel quality in the time and frequency domains. The channel status can be obtained by measuring the received transmission power of the reference signals sent on the DL [DPSB08].

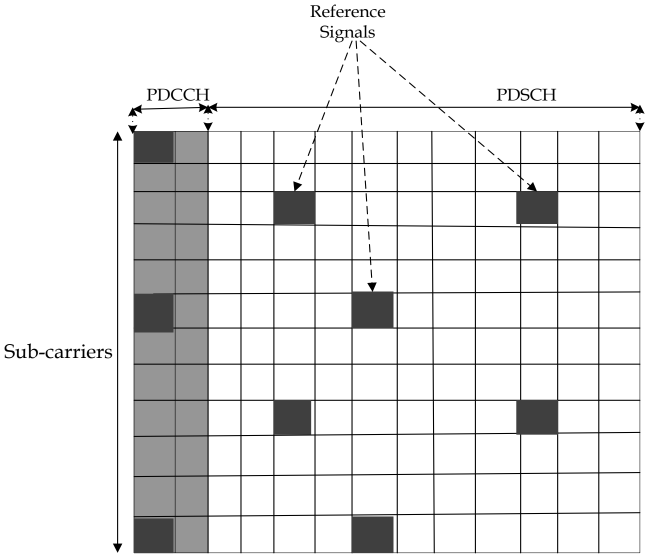


Figure 2-7 Reference signals

According to 3GPP standardisation, reference signals are embedded in the PRB as shown in Figure 2-7. Based on the channel-status reports, the DL scheduler can assign resources for downlink transmission to different mobile terminals. In principle, a scheduled terminal can be assigned an arbitrary combination of 180k Hz wide resource block in each 1 ms TTI [DPSB08].

During each TTI the eNB scheduler shall [3GPPLTE12d]:

- **Consider the physical radio environment conditions per User Equipment (UE):** All UEs report their perceived radio quality, as an input to the scheduler to decide which modulation and coding scheme to use. The solution relies on rapid adaptation to channel variations, employing Hybrid Automatic Repeat Request (HARQ) with soft-combining and rate adaptation.

- **Prioritise the QoS service requirements amongst the UEs:** LTE supports both delay sensitive real-time services as well as data services that require high peak data rates. It prioritises UEs according to their service requirements, to improve QoS provision.

- **Inform the UEs of allocated radio resources:** The eNB schedules the UEs both on the DL and on the UL. For each UE scheduled in a TTI, a Transport Block (TB) is generated carrying user data, which is delivered on the transport channel. The scheduled users are informed about their allocated resources before sending their data. This information is called scheduling control information, which is sent on control channel [3GPPLTE12d].

2.5 QoS Classes and QoS Requirements

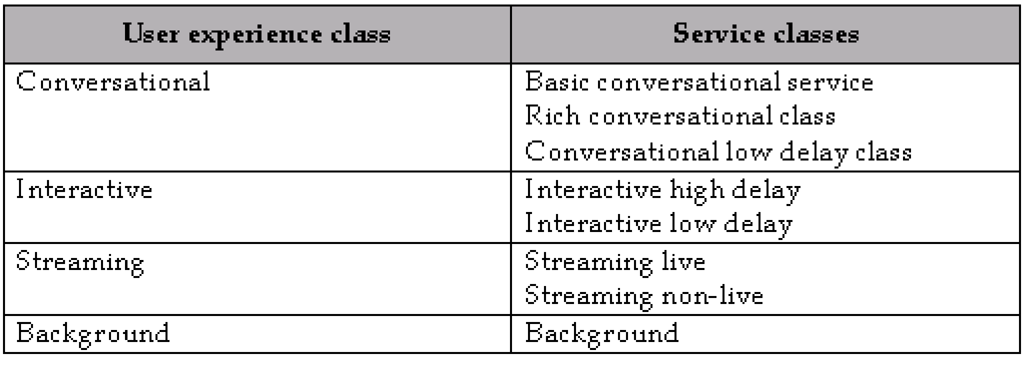
From the end user's point of view, the QoS of the service that the user has requested is perceived by the user's experience in relation to a particular application. For example, a web browsing user perceives QoS mainly in terms of the time it takes until a webpage is fully displayed after clicking on a hyperlink or entering a URL. From the technical point of view, this duration results from a complex interaction of factors like throughput, packet delay, and residual bit error ratio. Similarly, the quality of a Voice over IP (VoIP) call is perceived by the end users in terms of delay and voice quality. Technically, the QoS of a VoIP call can be expressed in packet delay and the residual bit error ratio.

In modern communication networks, there are more and more emerging services with different QoS requirements which can be technically represented by different sets of QoS parameters. This motivates the introduction of user experience classes that group together services with similar QoS attributes. In [3GPP03] recommendation, this classification is as:

- conversational class,
- interactive class,
- streaming class,
- background class

In LTE-A, the conversational class includes basic conversational service, rich conversational service and conversational low delay service as given in Table 2-1 [3GPP09c] [3GPP12a]. In addition, it also supports interactive high and low delay service, streaming live and non-live and background service.

Table 2-1 Service Classes



User experience class	Service classes
Conversational	Basic conversational service Rich conversational class Conversational low delay class
Interactive	Interactive high delay Interactive low delay
Streaming	Streaming live Streaming non-live
Background	Background

The basic conversational service class comprises basic services that are dominated by voice communication characteristics. The rich conversational service class consists of services that mainly provide synchronous communication enhanced by additional media such as video, collaborative document viewing, etc. Conversational low delay class comprises real-time services that have very strict delay requirements. In the interactive user experience class two service classes are distinguished. Interactive services that permit relatively high delay which usually follow a request-response pattern (e.g. web browsing, database query, etc.). In such cases, response times in the order of a few seconds are permitted. Interactive services requiring significantly lower delay is remote server access or remote collaboration. In the streaming user experience class there are two service classes. The differentiating factor between those two classes is the live or non-live nature of the content transmitted. In case of live content, buffering possibilities are very limited, which makes the service very delay-sensitive. In the case of non-live (i.e. pre-recorded) content, layout buffers at the receiver side provide a high robustness against delay. The background class only contains delay-insensitive services, so that there is no need for further differentiation [Jayakumari10].

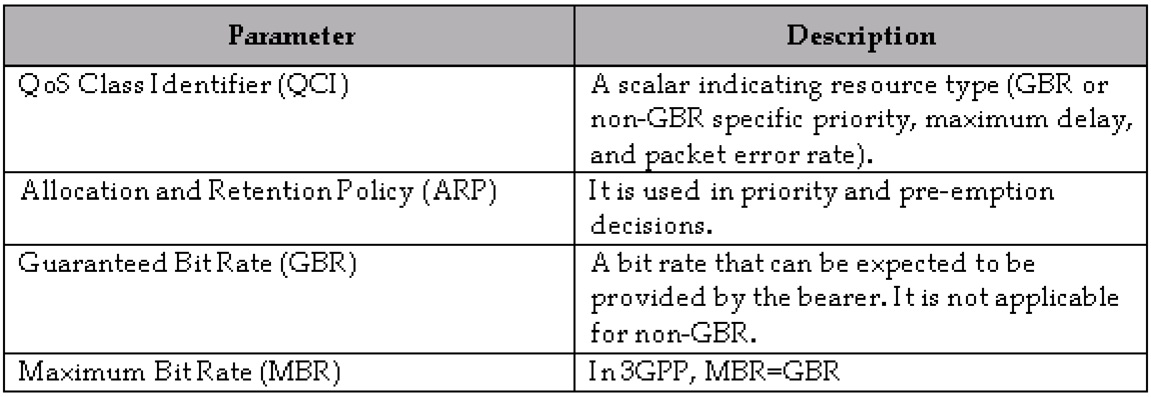
Since classification of the services listed in Table 2-1 is based on user experience class and service classes that are similar in terms of required QoS. The detailed services that are combined into above mentioned service classes in Table 2-1 are represented in the Table 2-2 [3GPP07].

Table 2-2 Service Classification

User experience class	Service class	Example services
Conversational	Basic conversation	Voice telephony (including VoIP), Emergency calling (call to public safety points), Push-to-talk (quick exchange of information).
	Rich conversation	Video conference, High-quality video telephony, Remote collaboration, e-Education (e.g. video call to teacher), Consultation (e.g. video interaction with doctor), Mobile commerce.
	low delay Conversation	Interactive gaming, Consultation, Priority service.
Interactive	Interactive high delay	e-Education (e.g. data search), Consultation (e.g. data search), Internet browsing, Mobile commerce (buying/selling through wireless handheld devices), Location-based services (to enable users to find other people, vehicles, resources, services or machines).
	Interactive low delay	Emergency calling, e-mail (Internet Message Access Protocol, IMAP server access), Remote collaboration (e.g. desktop sharing), Push alerting (e.g. notification of hazardous situation), Messaging (instant messaging), Mobile broadcasting/multicasting (mobile interactive personalised TV), Interactive gaming.
Streaming	Streaming live	Emergency calling, Push alerting, e-Education (e.g. remote lecture), Consultation (e.g. remote monitoring), Machine-to-machine (e.g. observation), Mobile broadcasting/multicasting, Multimedia.
	Streaming non-live	Mobile broadcasting/multicasting, e-Education (e.g. education movies), Multimedia, Mobile commerce, Remote collaboration.
Background	Background	Messaging, Video messaging, Public alerting, e-mail (transfer Receiver /Transmitter, e.g. Post Office Protocol, POP), Machine-to-machine, File transfer/download, e-Education (file upload/download), Consultation (file upload/download), Internet browsing, Location-based service.

The QoS framework of LTE-A builds upon the one developed for LTE and allows the support of wide range of services as described above. An end-to-end, class based QoS architecture has been defined by LTE where a bearer is the level of granularity for QoS control. It is much simpler than 3G networks and enables bearers to be mapped to a limited number of discrete QoS classes. All network nodes can be pre-configured for these classes limiting the QoS information which is needed to be dynamically signalled. The user-level standardised QoS parameters are QoS Class Identifier (QCI), Allocation and Retention Policy (ARP), Guaranteed Bit Rate (GBR), and Maximum Bit Rate (MBR). They are given in Table 2-3 along with their brief description [3GPP12c].

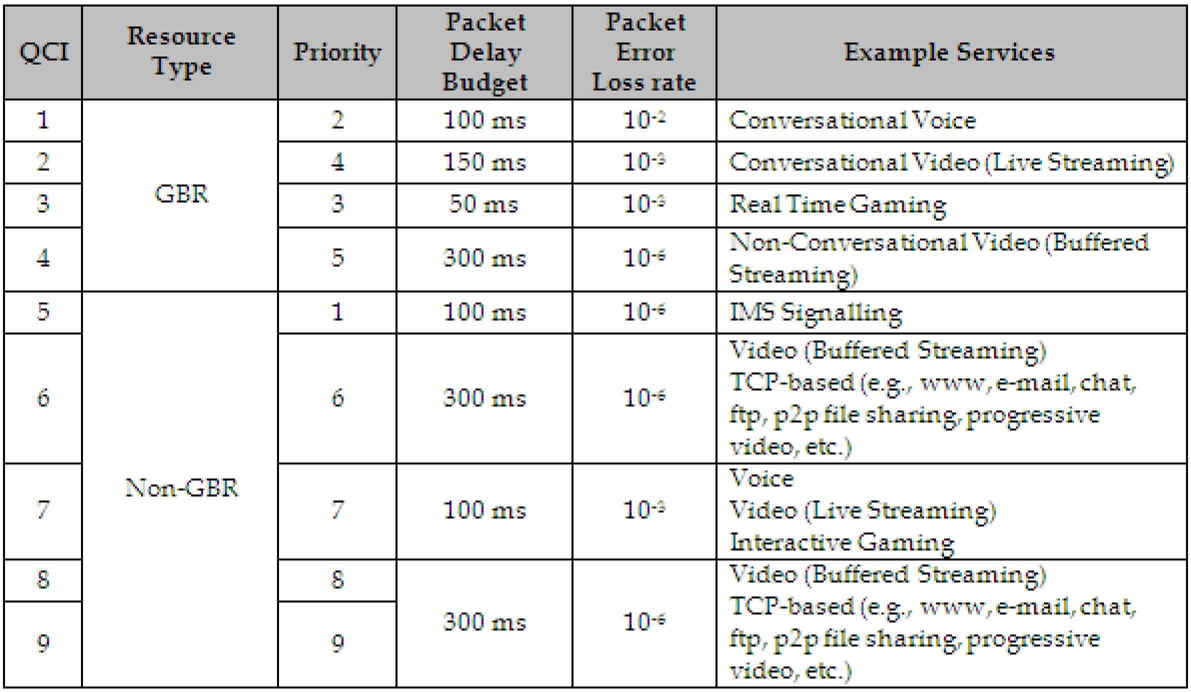
Table 2-3 Bearer QoS Profile



Parameter	Description
QoS Class Identifier (QCI)	A scalar indicating resource type (GBR or non-GBR specific priority, maximum delay, and packet error rate).
Allocation and Retention Policy (ARP)	It is used in priority and pre-emption decisions.
Guaranteed Bit Rate (GBR)	A bit rate that can be expected to be provided by the bearer. It is not applicable for non-GBR.
Maximum Bit Rate (MBR)	In 3GPP, MBR=GBR

QCI is used as a reference to access node-specific parameters that control bearer level packet forwarding treatment (e.g. scheduling weights, admission thresholds, queue management thresholds, link layer protocol configuration, etc.), and that have been pre-configured by the operator owning the eNB. The goal of standardising a QCI with corresponding characteristics is to ensure that applications/services mapped to that QCI receive the same minimum level of QoS in multi-vendor network deployments [3GPP12c]. A standardised QCI and corresponding characteristics is independent of the UE's current access (3GPP or Non-3GPP). A one-to-one mapping of standardised QCI values to standardised characteristics is for instance captured in [3GPP11b] for LTE. The configuration of those QoS parameters allows LTE-A to support a wide range of services. Table 2-4 gives the QoS parameters and priorities of different traffic types defined by 3GPP [3GPP12a].

Table 2-4 LTE QoS Class Identifiers and Service Priority



QCI	Resource Type	Priority	Packet Delay Budget	Packet Error Loss rate	Example Services
1	GBR	2	100 ms	10^{-2}	Conversational Voice
2		4	150 ms	10^{-3}	Conversational Video (Live Streaming)
3		3	50 ms	10^{-3}	Real Time Gaming
4		5	300 ms	10^{-6}	Non-Conversational Video (Buffered Streaming)
5	Non-GBR	1	100 ms	10^{-6}	IMS Signalling
6		6	300 ms	10^{-6}	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
7		7	100 ms	10^{-3}	Voice Video (Live Streaming) Interactive Gaming
8		8	300 ms	10^{-6}	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
9		9			

The QCI, as mentioned earlier, is a scalar that maps to a set of characteristics describing the expected packet forwarding treatment. The GBR resource type provides the required GBR while non-GBR does not provide any specific guarantee in terms of bit rate e.g. background traffic. Priority 1 corresponds to the highest priority used to differentiate bearers in case of resource shortage. Packet Delay Budget (PDB) is a soft upper bound with a confidence level of 98% for a time that a packet may be delayed between the gateway and UE [GPKM08]. The QCI 6, 8 and 9 apparently look similar however the QCI 6 can be used only if the network supports Multimedia Priority Services (MPS), the QCI 8 is used for dedicated premium bearer for any subscriber or group of subscribers and the QCI 9 is typically used for the default bearer of a UE/PDN for non-privileged subscribers. More details on the QCI can be found in [3GPP12a].

Figure 2-8 shows the QoS framework of LTE which illustrates the basic operation of a QoS-aware MAC Scheduler in the DL direction [RokeLTE13].

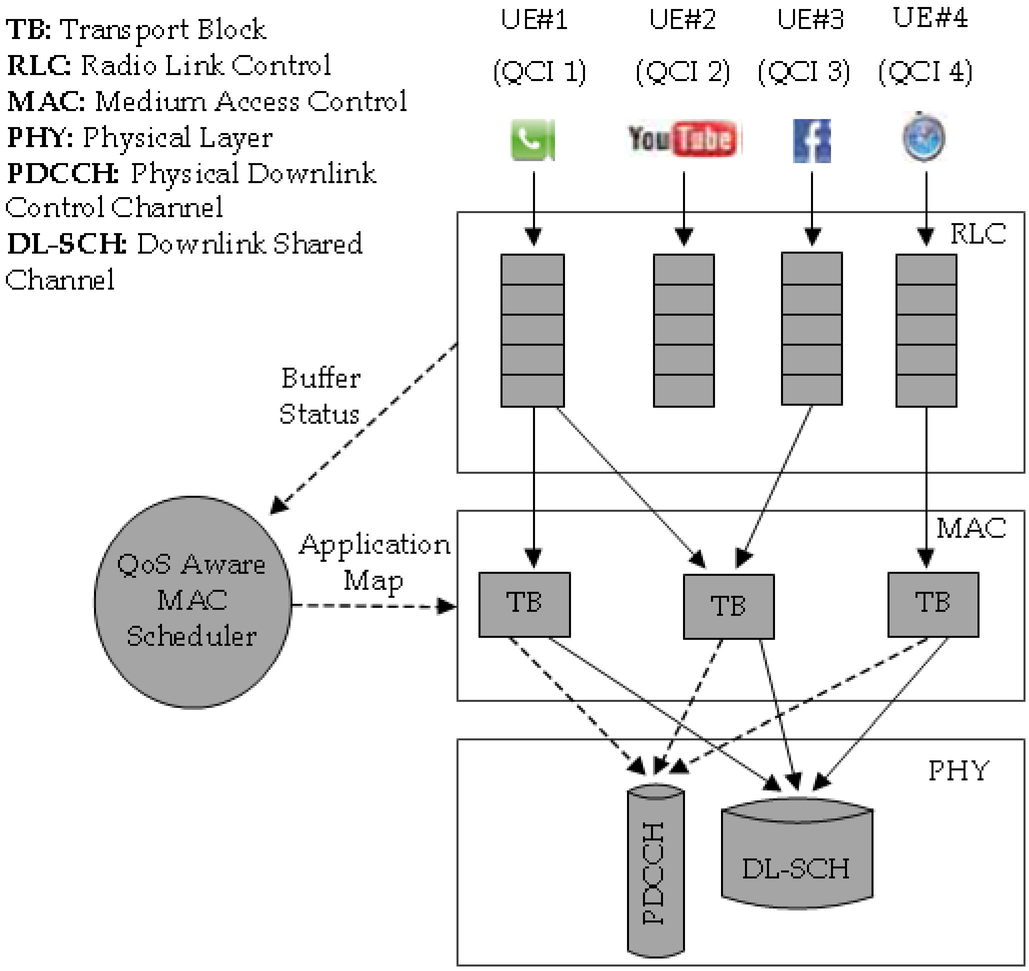


Figure 2-8 QoS framework of LTE network

Data for multiple services is queued in the Radio Link Control (RLC) sub-layer and the MAC Scheduler receives buffer status updates as new data arrives. The MAC Scheduler determines the DL allocation of the radio resource for each sub-frame. These allocations are signalled to the MAC sub-layer which constructs the UE specific TB. Each TB contains data from one or more service classes.

2.6 Classic Packet Scheduling Algorithms

In this section, three classic packet scheduling algorithms RR, MAX C/I and PF are described.

2.6.1 Round Robin (RR) Algorithm

The RR algorithm schedules users cyclically allocating a fair share of the available radio resource to all users [TCT10]. In the wireless networks this algorithm is capable to achieve starvation free scheduling. However the overall system throughput becomes low because RR algorithm does not take into account of wireless channel conditions and cannot exploit MU diversity. In general, MU diversity gain arises from the fact that in wireless systems with a number of users, the utility value i.e. achievable data rate of a given resource block varies from one user to another. This is because users have different channel conditions and their achievable data rate varies [DPSB08]. These variations allow the overall system performance to be maximised by allocating each resource block to the user that can best exploit it [TCT10]. To illustrate MU diversity gain, an example of single cell OFDMA system with two users is taken where users are scheduled by RR algorithm.

In this example, following assumptions are taken into account of.

1. Users' channel responses are independent.
2. Users have perfect knowledge of channel state information.
3. There is perfect feedback from each user to the eNB.
4. The eNB gathers channel measurements from each user.

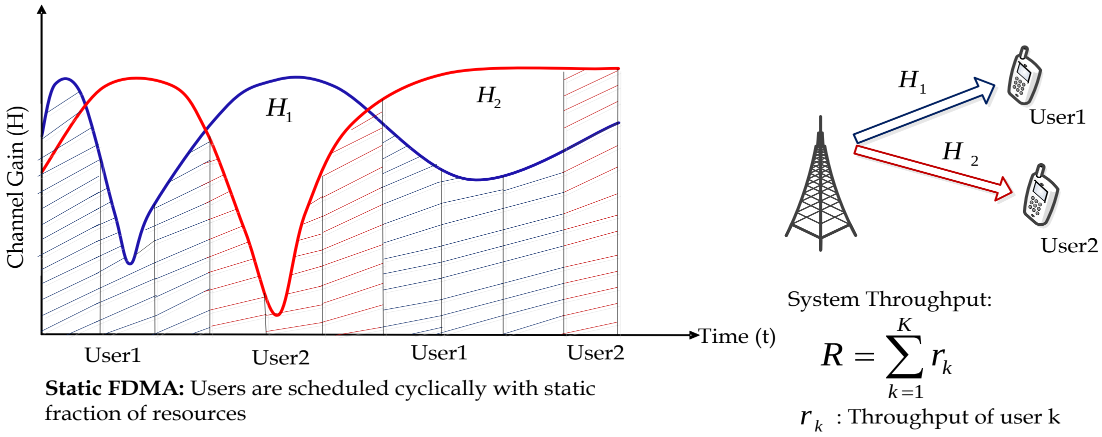


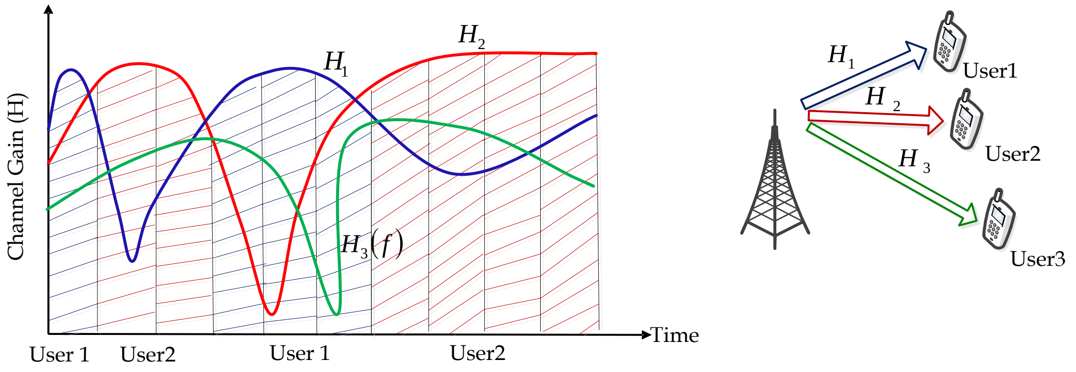
Figure 2-9 Round Robin (RR) algorithm

Figure 2-9 shows the principle of RR algorithm in a wireless scenario. The channel gain of two users is represented by the curves and the shaded area under the curves represents the allocation of the available radio resource to these users. H_1 (in blue) and H_2 (in red) represent the channel gain of user 1 and user 2, respectively. These two users are scheduled cyclically to allocate a fair share of the available radio resource. As a result, fairness among users is high but throughput achieved by these users is low as RR algorithm does not consider channel conditions of users while taking scheduling decision. As the overall system throughput R is the sum of the achieved throughput for all active users in the system, in this particular case, the sum of user 1 and user 2, thus it is related to the achieved throughput of each user. To each user, this achievable throughput is directly related to its channel gain [DPSB08]. A user with good channel conditions achieves good throughput which results in good system throughput. If in each scheduling decision, users with good channel condition are scheduled, the overall system throughput and system spectral efficiency can be increased significantly [DPSB08]. This is the essence of the Maximum Carrier to Interference (MAX C/I) Algorithm, described in the next section.

2.6.2 Maximum Carrier to Interference (MAX C/I) Algorithm

MAX C/I algorithm takes channel dependent scheduling decisions [PJNTTM03]. It basically deals with the question of how to share available radio resources among users having different channel conditions to achieve as efficient resource utilisation as possible [DPSB08]. It tries to exploit the channel variations through appropriate processing prior to transmission of data. To increase system throughput (or system spectral efficiency), users are scheduled when they have good channel conditions.

The MAX C/I algorithm schedules users with good channel conditions [HT09] [PPMKRKM07]thus maximising the system throughput and thus system spectral efficiency. In other words MAXC/I algorithm exploit MU diversity to maximise system spectral efficiency. Figure 2-10 shows the scheme of MAX C/I algorithm.



Dynamic OFDM: A user with the highest capacity is scheduled (exploitation of MU diversity)

Figure 2-10 Maximum Carrier to Interference (MAX C/I) algorithm

A system of three active users is considered in Figure 2-10 in which three curves show the channel gain of three users and the shaded area under the curves shows resource allocation to these users. H_1 (in blue), H_2(in red) and H_3(in green) represent the wireless channel gain of user 1, user 2 and user 3, respectively. A user is allocated radio resource only when it has good channel conditions, as shown. User 1 and user 2 have good channel condition and are scheduled but user 3 is never scheduled because it always has bad channel condition. As

MAX C/I algorithms schedules users only when they have good channel conditions, system spectral efficiency is significantly increased as compared to the RR algorithm.

Generalising, a system consisting of K number of users where $K = k_1,\ k_2,\ k_3, \ldots\ldots K_k$ and M number of available PRBs, where $M = m_1, m_2, m_3, \ldots\ldots M$, the priority metric $P_{k,m}(t)$ for MAX C/I algorithm is given by Equation 2-1 [JNTMM08] [AKRSW01].

$$P_k(t) = argmax_{k,m}\left(r_{k,m}(t)\right) \tag{2-1}$$

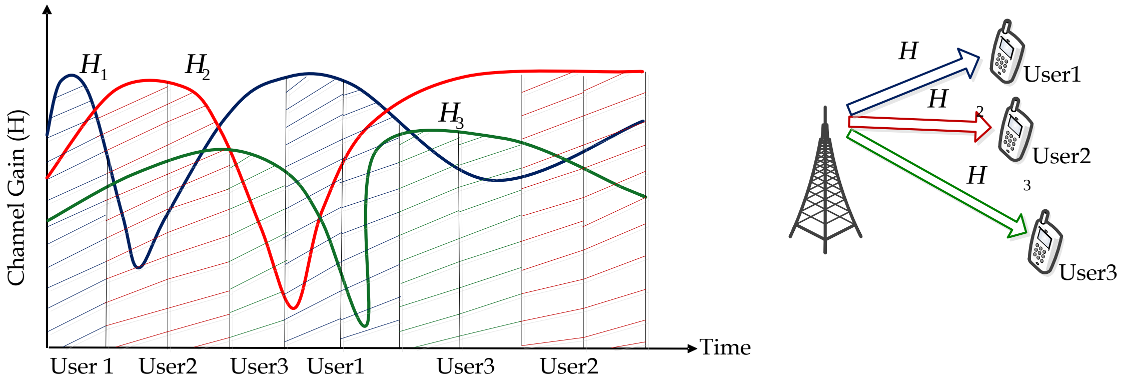
In which $\mathrm{r_{k,m}(t)}$ is maximum instantaneous supportable data rate of user k at time t.

However in MAX C/I algorithm users with bad channel conditions, for example user 3 (in green) in Figure 2-10, suffers starvation, which is not fair. The MAX C/I algorithm thus achieves very low fairness as compared with RR algorithm.

A trade-off is needed between system spectral efficiency and user fairness to achieve an efficient resource allocation. The PF algorithm is developed to make a good trade-off between system spectral efficiency and user fairness [GBP08] [WXZXY03].

2.6.3 Proportional Fairness (PF) Algorithm

The PF algorithm takes both fairness among users and system spectral efficiency into consideration and allocates the radio resource to users based on the ratio of their achievable instantaneous throughput and their time averaged throughput [PBR05]. It allocates a fair share of the radio resource to all users and maintains good system spectral efficiency at the same time, by considering the trade-off between user fairness and system spectral efficiency.



Dynamic OFDMA: A user with maximum ratio of achievable throughput to the mean throughput is scheduled

Figure 2-11 Proportional Fairness (PF) algorithm

Figure 2-11 shows the scheme used in the PF algorithm by considering a system with three active users. When compared with Figure 2-10 (with MAX C/I algorithm) where user 3 (in green) has not been scheduled at all, PF algorithm allocates a fair share of radio resource to user 3 as well. The PF algorithm thus, shows a good trade-off between system spectral efficiency and user fairness by achieving fairness higher than MAX C/I and spectral efficiency higher than RR algorithm.

The PF scheduling priority metric P_k(t) is given in Equation 2-2 [JNTMM08] [GBP08].

$$P_{k,m}(t) = \frac{r_{k,m}(t)}{R_k(t)} \qquad \text{at any time slot} \qquad (2\text{-}2)$$

In which R_k(t) is the average achieved throughput of user k at time t which is updated by the following Equation 2-3 [AZXY03].

$$R_k(t+1) = \left(1 - \frac{1}{t_c}\right) R_k(t) + \frac{1}{t_c} \sum_{k,m}^{M} r'_{k,m}(t) \qquad (2\text{-}3)$$

In which t_c is the length of time window to calculate the average throughput, $1/t_c$ is called the attenuation co-efficient with the widely used value 0.001, $r'_{k,m}(t)$ is the acquired data rate of user k at PRB m if m is allocated to user k at time t.

2.7 Generalised Proportional Fairness Algorithm

The traditional PF scheduler allocates the user who maximises the ratio of achievable instantaneous data-rate over average achieved data-rate [LL05]. The PF approach is broadened to Generalised Proportional Fairness (GPF) algorithm [CJAN05]where new weighting factors are introduced to the conventional PF algorithm. Referring to Equation 2-2, GPF algorithm can be expressed as in Equation 2-4.

$$P_k(t) = \frac{[r_{k,m}(t)]^a}{[R_k(t)]^b} \tag{2-4}$$

In Equation 2-4, by changing the values of parameters a and b, the trade-off between spectral efficiency and fairness can be controlled. For a parameter setting $a = b = 1$ conventional PF scheduling is achieved and tuning between these parameters, the trade-off between fairness and throughput can be tweaked. Increasing a will increase the influence of achievable instantaneous data-rate which enhances the probability of a user in currently good condition to be scheduled. This results in higher system spectral efficiency, but lower fairness. Increasing b will increase the influence of the average data rate $R_k(t)$ which increases the probability of a user with a low average data rate to be scheduled thus increasing fairness level of users.

2.8 QoS Aware Packet Scheduling Algorithms

The classic algorithms focus on fairness (RR algorithm), system spectral efficiency (MAX C/I algorithm) or a trade-off between fairness and system spectral efficiency (PF algorithm). However in the LTE-A networks, which aims to support diverse applications with variety of QoS requirements, apart from system spectral efficiency and user fairness, the crucial point is to meet users′ QoS requirements in a multi-service mixed traffic environment. For example real-time services like audio phone and video conference require end-to-end performance guarantees because a reliable and timely transmission is needed. On the other hand non-real time services can tolerate delays to a certain limit but require long-term minimum throughput guarantees.

The classic packet scheduling algorithms (section 2.2) cannot achieve the set targets by International Telecommunication Union Recommendations (ITU-R) as they are only designed to improve fairness, spectral efficiency or a trade-off between them.

The following section presents some of the state-of-the-art on QoS aware packet scheduling algorithms followed by the motivation of the research work in this thesis.

2.8.1 Quality of Service (QoS) and Queue State Information (QSI)

QoS is defined as the ability of a network to provide a service to an end user at a given service-level where the service-level corresponds to end user experience such as packet delay or data rate [SLC06]. QoS aware packet scheduling algorithms focus on meeting QoS demands by using information on for example, channel and queue state.

In mixed traffic scenarios, QSI becomes important in addition to CSI [AZXY03]. It can make scheduling decision even more efficient especially in the next generation of mobile communication systems which support diverse range of applications. Typically this implies to minimise the amount of resources needed per user and thus allows for as many users as possible in the system, while still satisfying whatever quality of service requirements that may exist [NH06]. Most of the QoS aware packet scheduling algorithms takes into account of both QSI and CSI to make scheduling decisions to support QoS guarantees to different service types.

2.8.2 Modified Largest Weighted Delay First (MLWDF)

Largest weighted delay first (LWDF) [SR00] is designed for the coexistence of real-time services with different delay bounds but it cannot guarantee the throughput demands for non-real time services [3CZWS10]. To provide QoS to different both real-time and non-real time services, Modified Largest Weighted Delay First (MLWDF) algorithm was proposed in [AKRS01]. It supports the guarantees of delay and minimum throughput requirements of real-time and non-real time services, respectively in the third generation wireless networks.

In each time slot t, it serves a queue that maximizes the following metric.

$$\gamma_j W_j r_j \tag{2-5}$$

In which $W_j(t)$ is the HOL packet delay for user queue j, $r_j(t)$ is the instantaneous throughput with respect to flow j, and γ_j are arbitrary positive constants, which embody the QoS requirement, and provide QoS differentiation between the flows. In case of non-real time service where the QoS requirement is minimum throughput instead of delay threshold, this algorithm is modified by conjunction with a token bucket control. With each queue a virtual token bucket is associated and tokens in the bucket i arrive at the constant arrival rate r_i which is equal to the minimum throughput requirement of non-real time service. The scheduling decision is taken by the same metric as for real-time service but $W_j(t)$ being not the delay of actual HOL packet of flow j but the delay of the longest waiting token in token bucket i. In this way MLWDF satisfies QoS of real-time and non-real time services in a TDM system. However MLWDF does not consider system-level performance such as system spectral efficiency and user fairness.

2.8.3 Subcarrier Allocation for OFDMA Systems

The MLWDF algorithm was extended to an OFDMA system in [PBA05]. Users are allocated radio resources based on the product of their channel conditions and delay of the HOL packet. Thus users with good channel conditions and longer waiting times are prioritised to transmit their data. The algorithm, thus, prioritises a queue j by the following metric.

$$j = argmax_i \left[\frac{\overline{a}_k(t)}{d_k(t)}\gamma_{k,m}(t)W_{k,m}(t)\right] \quad k = 1,2, \ldots\ldots K \tag{2-6}$$

In which $\overline{a}_k(t)$ and $\overline{d}_k(t)$ are mean arrival and mean throughout respectively for flow j averaged over a time period, $\gamma_{k,m}$ is channel gain of user k on subcarrier m and $W_k(t)$ is delay of HOL packet of user k. In this paper the delay of HOL is updated when all the subcarriers are allocated which is not an accurate way of characterising the whole queue status during a scheduling loop as it is exploited in MLWDF. Furthermore, the HOL delay cannot be directly used for QoS guarantee of non-real time service as it does not embody throughput requirements of it.

2.8.4 Sum Waiting Time Based Scheduling (SWBS)

In [SYLX09], a Sum Waiting Based Scheduling algorithm (SWBS) is presented which updates the HOL delay after each subcarrier allocation instead of after allocating all subcarriers, to make resource allocation accurate and fair as in [AKRS01]. Packet waiting time is used as a metric for the satisfaction of real-time and non-real time services. However for non-real time service, it is taken as virtual arrival time related to the minimum throughput of non-real time service.

A PRB m is allocated to a user k based on the following metric.

$$[k,m] = argmax_{k,m\in M}\left(W_k(t) \times \left[H_{k,m}(t)\right]^2\right) \tag{2-7}$$

In which $W_k(t)$ is waiting time for user k during time t and $\left[H_{k,m}(t)\right]^2$ is channel gain of user k on subchannel m during time t.

The waiting time for RT user is,

$$w_{k,i}^{RT} = t - T_{k,i}^{arrival} \tag{2-8}$$

In which $W_{k,i}^{RT}$ is the waiting time of i^{th}packet of the k^{th} user at time t, $T_{k,i}^{arrival}$ is arrival time of packet i of user k. The waiting time is summed up for all packets of a user and normalised by delay budget (DB) of real-time service as given in Equation 2-9.

$$W_k^{RT}(t) = \frac{1}{DB_k^{RT}} \sum_{i=1}^{q_k(t)} w_{k,i}^{RT} \tag{2-9}$$

In which $q_k(t)$ is the queue length of user k. For non-real time service, waiting time of users is calculated by considering virtual arrival time of users which is related with minimum throughput requirements of non-real time service. This algorithm improves QoS to real-time and non-real time services. However the system-level performance in terms of system throughput and user fairness is not considered in the design of this algorithm.

2.8.5 Mixed Traffic Packet Scheduling in UTRAN (MIX)

In [JNTMM08], packet scheduling of mixed traffic in UTRAN LTE downlink is presented, in which scheduling decisions are taken both in the time and frequency domains. The scheduling unit is called a scheduling object, which corresponds to one UE with the knowledge of traffic class, buffer sizes etc. The classifier divides different scheduling objects into different traffic class dependent queues, TD scheduler chooses a certain number of objects for scheduling candidates and FD scheduler does the actual PRB allocation. It uses conventional algorithms (RR, MAX C/I and PF) to sort users in different queues. TD scheduler uses either strict priority or fair scheduling algorithms in the TD scheduler to select a pool of users from the head(s) of queue(s). In strict priority TD scheduling, queues are emptied from top to bottom and in TD fair scheduling one user is picked from each queue at a time. FD scheduler allocates PRBs to the candidate UEs chosen by the TD scheduler. The scheduling scheme presented here is good however to improve QoS provision instead of using conventional algorithms, service specific queue sorting algorithms can be used. For example queuing delay aware sorting algorithms can be used for VoIP, while the background could use e.g. proportional fairness algorithm.

2.9 Machine Learning in Scheduling

To enhance the scheduling performance in different domains, Artificial Intelligence (AI) can be integrated in communication networks. The AI technologies offer many new and exciting possibilities for the next generation of mobile communication networks [JN87]. Machine learning is a branch of AI which deals with the study of systems that can learn from data. For example, a machine learning system can be trained on email messages to learn to distinguish between spam and non-spam messages. And after this learning, it can be used to classify new emails into spam or non-spam folders.

In scheduling, data relevant to different parameters such as average PDR can be used to train the system for the next scheduling decisions. As for example in this thesis, data on average PDR of real-time service in previous TTIs is used as input in the Hebbian learning process to train the system for taking adaptive scheduling decisions. Similarly average PDR data of individual RT users is used in K-mean clustering algorithm to train the system to prioritise users with higher average PDR values.

2.9.1 Hebbian Learning

One of the learning procedures for two-layer networks is the Hebbian Learning rule, which is based on a rule initially proposed by Hebb in 1949. The Hebbian postulate of learning discovered by Hebb [Hebb49] is one of the oldest learning rules. It states how much the weight of the connection between two units should be increased or decreased in proportion to the product of their activation [Hebb49] [WA93] [RM86]. The learning process extracts information from environment and stores it in synaptic weights. Therefore at each step of the learning process, an appropriate adjustment is applied to each synaptic weight. This adjustment is represented for synaptic weight w_j in the nth step by $\Delta w_j(n)$, i.e.

$$w_j(n+1) = w_j(n) + \Delta w_j(n) \tag{2-10}$$

In which $\Delta w_j(n)$ may have positive or negative values. A synapse that obeys Hebbian learning rule is called a Hebbian synapse. A precise definition of Hebbian synapse is proposed in [Haykin08]:

> *A synapse uses a time-dependent, highly local and strongly interactive mechanism to increase synaptic efficiency as a function of the correlation between the presynaptic and postsynaptic activities.*

Hebbian learning process is adopted in this thesis because its computation time is smaller as compared to other learning rules such as regressive learning and learning tree. It takes decisions by extracting information from the environment and comparing current and historical values of the extracted information. As the scheduling intervals are smaller and quick decisions are needed in each interval, Hebbian learning process is thus appropriate to adaptively allocate the available radio resource.

Previously, Hebbian learning rule was used in [KH10] for dynamic spectrum management in Cognitive Radio (CR) to estimate the presence of primary users (PUs) in the environment; PUs is the licensed users and allowed to operate in the spectrum band bought by the wireless service provider. It helps in preventing collisions of CR units with PUs.

2.9.2 Clustering

Clustering is a grouping technology, which partition data into a number of clusters/groups [XW05]. To deal with the variable QoS requirements of different users of a service type, clustering is a suitable technique to integrate in scheduling algorithms to rearrange users' priority according to their QoS requirements. More specifically, clustering is a technique to group together a set of items having similar characteristics [SCDT00]. In scheduling domain, where there is always a need to change priorities of different users according to their QoS requirements, clustering is a good technique to construct groups of users with similar QoS requirements. It helps in setting priorities of users by sorting these groups in an appropriate way.

K-mean clustering algorithm is the most commonly used clustering algorithm, which is adopted in many works [PKVP06] [PSPP07]]. It is incorporated in the proposed scheduling architecture to improve the performance of real-time voice service.

K-mean clustering algorithm is used for partitioning (or clustering) N data points into Q disjoint subsets S_j containing N_j data points so as to minimise the sum-of-squares criterion.

$$J = \sum_{j=1}^{Q} \sum_{n \in S_j} \left| x_n - \mu_j \right|^2 \qquad (2\text{-}11)$$

In which, x_n is a vector representing the nth data point and μ_j is the geometric centroid of the data points in S_j.

The algorithm consists of a re-estimation procedure to group the given data into groups. Initially, the data points are assigned at random to the Q sets. For step 1, the centroid is computed for each set. In step 2, every point is assigned to the cluster whose centroid is closest to that point. These two steps are alternated until a stopping criterion is met, i.e., when there is no further change in the assignment of the data points.

Clustering has already been used in many domains especially on web, such as aiming at improving web applications [SCDT00] [PKVP06]. Clustering based scheduling gives variable priorities to different users belonging to the same service type. For example in [PSPP07], a clustering based scheduling algorithm is used to organise users of a network into groups based on the number of their requests per channel. The transmission priority then starts from the group with the highest requests. It improves network performance in terms of higher network throughput while keeping mean packet delay at lower levels as compared to the conventional scheduling algorithms [PSPP07].

2.10 **Problem Description**

Packet scheduling algorithms described above and in other existing literature are designed to improve scheduling performance in different domains separately. For example they either consider system-level scheduling performance i.e. system spectral efficiency and user fairness or the user-level scheduling performance i.e. QoS of real-time and non-real time services. However, at the time of this thesis being written, the joint consideration of user-level and system-level performance of QoS aware packet scheduling has not been researched in depth. As for example [JNTMM08] [LL05] focuses on performance enhancement on fairness level and system spectral efficiency but lacks QoS support to different services. And the scheduling schemes in [SR00] [3CZWS10] [AKRS01] [PBA05] [SYLX09] mainly address the support of QoS to real-time and non-real time services but do not investigate the system–level performance in terms of spectral efficiency and fairness.

As in a MU environment, the radio resources are shared among multiple users. A wireless network must be able to serve a diverse set of users in a highly dynamic, resource constrained and data intensive environments [PR80]. In a large scale wireless network, user's channel conditions, queue status and traffic type requirements are highly diverse. Therefore each radio resource in the time-frequency dimension is likely to have high utility value to certain users. Consequently, the total channel capacity or spectral efficiency of the network can be significantly increased through effective resource allocation in MAC layer. This motivates the design of highly adaptive MAC layer protocols and algorithms that can cope with the channel and traffic dynamics [KSA08], to achieve better resource allocation.

The work in this thesis addresses the scheduling problem in a multi-service mixed traffic environment for the DL transmission of LTE-A networks where the competition to get radio resources is very high and there are strict QoS requirements. In such environment, the crucial point is to clearly define the QoS requirements of different traffic types, their demands for radio resources, their channel conditions and queue status to support their demands. Combined consideration of this information can lead to a more efficient packet scheduling algorithm in terms of both the user-level performance and the system-level performance. An intelligent packet scheduling architecture can achieve this goal by making scheduling decisions adaptive to the network scenarios and also adaptive to the achieved performance of different services measured by QoS parameters.

2.11 **Summary**

In this chapter background knowledge of radio access technologies, packet scheduling, packet scheduling in OFDMA-based LTE-A networks, classic packet scheduling algorithms, QoS aware packet scheduling and machine learning based scheduling is presented. The state-of-the-art QoS aware algorithms are also discussed followed by the problem formulation which leads to the motivation of the research work in this thesis. The detail of the proposed research work is given in the next chapter, which includes the description on all novel algorithms presented in this thesis.

Chapter 3 Cross-Layer Design for QoS Aware Scheduling Architecture

This chapter presents the proposed cross-layer packet scheduling architecture and all novel algorithms used in it are described in detail. The concept of cross-layer design and the system model used in this thesis are explained comprehensively. At the end, it gives system-level performance evaluation parameters with their short description.

3.1 Principle of Design of Scheduling Architecture

A schematic diagram of the proposed Cross-Layer QoS Aware Packet Scheduling Architecture (CLPSA) is shown in Figure 3-1. It divides the scheduling process of the Time Domain (TD) and Frequency Domain (FD) into three stages; the *Traffic Differentiator*, the *TD Scheduler* and the *FD Scheduler.* These stages serve different functionalities by utilising information from different layers and work together to achieve better QoS for different services (user-level performance), and to achieve a good trade-off between the user-level and the system-level performance.

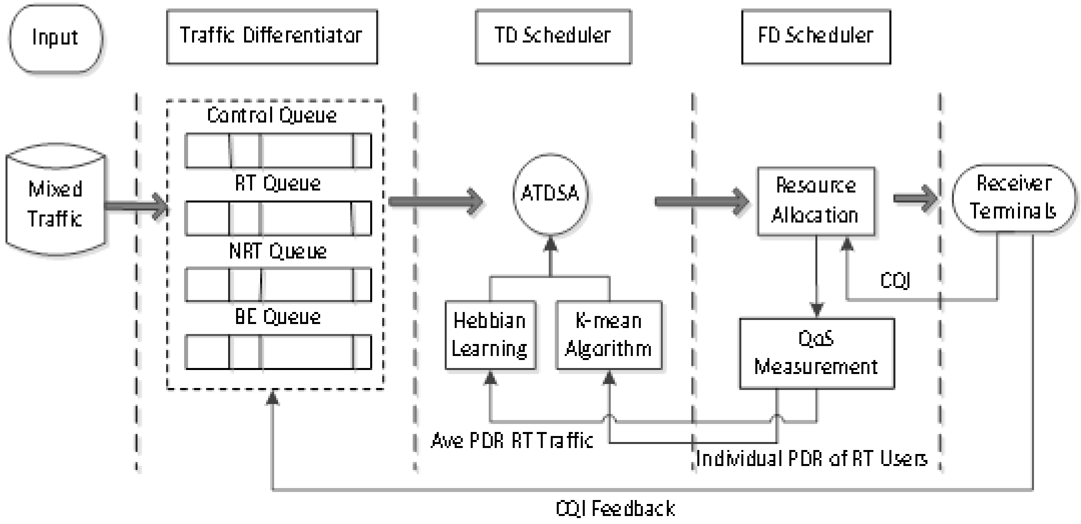
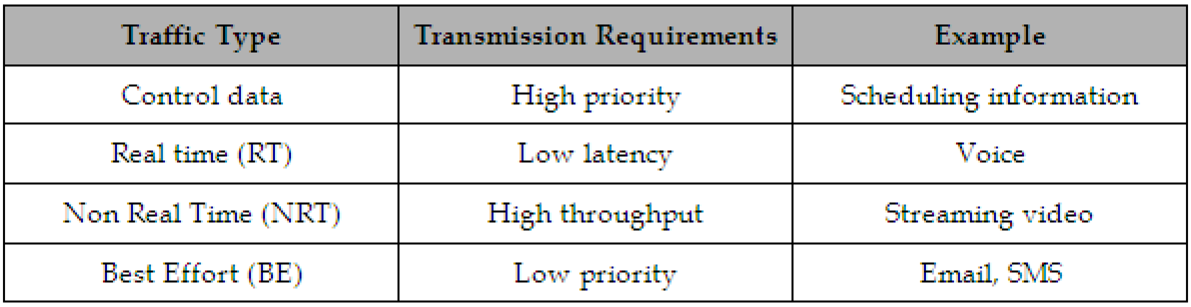


Figure 3-1 Cross-layer QoS aware packet scheduling architecture

3.1.1 Traffic Differentiator

The combination of different service requests of users in a mobile network can be seen as a mixed-traffic flow coming into eNB with a variety of QoS demands. The *Traffic Differentiator* segregates users requesting different services into service-specific queues and then sorts them with different scheduling priorities in the queues. In each queue, it uses a queue sorting algorithm which is suitable for the relevant service type in terms of QoS provision. In this thesis, the mixed traffic is differentiated into four queues: Control queue, RT queue, NRT queues and Best Effort (BE) queue consisting of control service, real-time voice service, non-real time streaming video service and background service. The queues are prioritised from top to bottom that is Control queue, RT queue, NRT queue and BE queue (Figure 3-1). These queues cover most of the common data types as illustrated in Table 3-1.

Table 3-1 Service classes



Traffic Type	Transmission Requirements	Example
Control data	High priority	Scheduling information
Real time (RT)	Low latency	Voice
Non Real Time (NRT)	High throughput	Streaming video
Best Effort (BE)	Low priority	Email, SMS

Control data is taken as one of the highest priority services, voice is taken as one of the low latency real-time services, streaming video is taken as one of the high throughput non-real time services and background service as one of the low priority services. Control data carries the scheduling information, which must be transmitted to the users before actual data transmission. Scheduling information consists of the scheduling grant or the PRBs allocated to users on which their data is sent. For a reliable communication, real-time voice service is very sensitive to delay requirements and non-real time streaming video requires a long-term minimum throughput guarantee for good quality of video streaming. Background service however does not have any QoS requirements, thus, comes in the category of low priority services. Control information of all scheduled users is transmitted in First Come First Serve (FCFS) manner. Two novel Service Specific queue Sorting Algorithms (SSSA) are proposed for real-time voice and non-real time streaming service. The users requesting background service are prioritised by using PF algorithm to maintain a good trade-off between fairness and system throughput.

3.1.2 TD Scheduler

The *TD Scheduler* makes decision on which users should be scheduled in the current TTI by considering both the available radio resource and the priorities of different users. An Adaptive Time Domain Scheduling Algorithm is proposed in the *TD Scheduler* stage, which incorporates Hebbian learning process and K-mean clustering algorithm, to achieve adaptive resource allocation. Hebbian learning is applied to decide how many RT, NRT and BE users should be scheduled in the current TTI. It makes a good balance of resource allocation among different service types in order to achieve a good system-level performance. K-mean clustering algorithm is used to further arrange the priorities of selected RT users based on their individual PDR in order to achieve better PDR fairness and reduced average delay of RT users.

3.1.3 FD Scheduler

The *FD Scheduler* makes the decision on the actual mapping of PRBs to the selected users from the *TD scheduler*. MU diversity is exploited in the frequency domain to allocate the best possible PRB to each selected user. Followed by the actual resource allocation decision, the QoS measurement unit calculates the average throughput of NRT users, average delay and average PDR of RT users and sends this information to the *TD scheduler* stage as part of the inputs for decision making in the next TTI. The CQI report from all users is assumed to be accurate and available to eNB in each TTI, as considered in most of the literature on channel aware scheduling.

3.1.4 Cross-Layer Concept

The cross-layer concept in the proposed architecture is used by taking information from different layers for making scheduling decisions, as shown in Figure 3-2. For each user, the *Traffic Differentiator* takes the traffic type information from the application layer, QoS requirements (e.g. delay budget or minimum throughput required) information from the network layer, the queue status information from the Radio Link Control (RLC) layer and the channel status information from the physical layer. The *TD scheduler* takes QoS measurements on average PDR from the MAC layer as part of the inputs for the Hebbian learning and the K-mean clustering algorithm. The *FD scheduler* makes the PRB mapping based on user priorities and per PRB CQI reports.

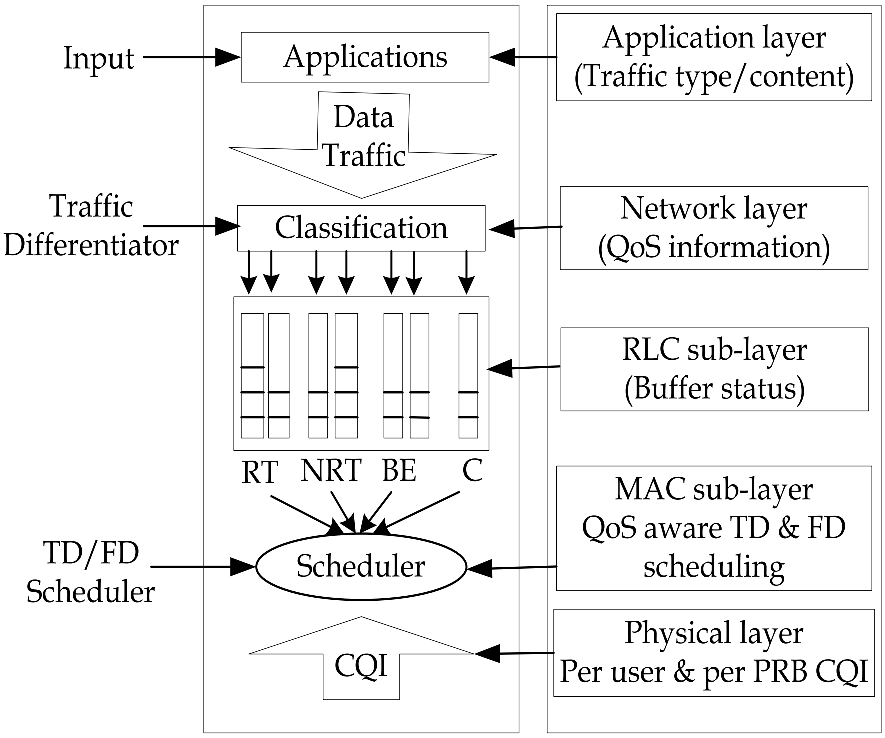

Input
Applications
Application layer (Traffic type/content)
Data Traffic
Traffic Differentiator
Classification
Network layer (QoS information)
RLC sub-layer (Buffer status)
RT
NRT
BE
C
MAC sub-layer QoS aware TD & FD scheduling
TD/FD Scheduler
Scheduler
Physical layer Per user & per PRB CQI
CQI

Figure 3-2 Cross-layer design for QoS aware scheduling architecture

3.2 **System Model**

This thesis considers the DL scheduling for a single cell OFDMA-based system where users are requiring various services with diverse QoS requirements in each TTI. There is one eNB located at the centre of cell, K number of total active mobile users and M number of total available PRBs. In this model, k represents the index of users, k=*1, 2,…..K,* and m represents the index of PRBs, m=*1, 2,….M.* The cross-layer design packet scheduling architecture presented in this thesis with novel SSSA and ATDSA can be applied on multi cell scenarios by using appropriate hand over and load balancing techniques.

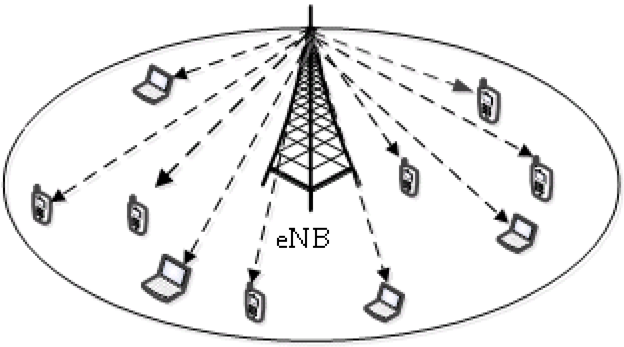


Figure 3-3 Downlink transmission

Figure 3-3 shows the DL transmission in a single cell system. The downlink channel is considered as a fading channel. The received signal at the mobile user k on PRB m at time t is $Y_{k,m}(t)$, and can be modelled as [SYLX09]:

$$Y_{k,m}(t) = H_{k,m}(t)X_{k,m}(t) + G_{k,m}(t) \tag{3-1}$$

In which $X_{k,m}(t)$ is the actual data received from eNB by user k on PRB m, $H_{k,m}(t)$ represents the channel gain of user k on PRB m and $G_{k,m}(t)$ is the complex Gaussian noise. The power allocation is assumed to be uniform on all PRBs as in [KKRHM08] and [SYLX09] and most of the literature on QoS aware scheduling.

The channel capacity of user k on PRB m can be expressed as [JL03]:

$$C_{k,m}(t) = Blog_2\left(1 + \frac{\left|H_{k,m}(t)\right|^2 p_{k,m}(t)}{\sigma^2 \Gamma}\right) \tag{3-2}$$

In which, B is the total available bandwidth, $C_{k,m}$ is the data rate that can be achieved by user k if allocated with PRB m, $p_{k,m}$ is the power allocated to user k at PRB m, σ^2 is the variance of Additive White Gaussian Noise (AWGN) and Γ is called Signal to Noise Ratio (SNR) gap and is a function of target Bit Error Rate (BER). For AWGN channel, it can be expressed as Γ, Γ = $ln\,(5 \times\ BER/1.5)$ [GS97].

Table 3-1 gives some important notations used in this thesis to explain algorithms at different stages of the proposed scheduling architecture.

Table 3-2 List of Notations

Notations	Description
$m \in 1,2, \dots M$	The m^{th} sub-carrier
$k \in 1,2, \dots K$	The k^{th} user
R_k	Average achieved throughput of user k
$C_{k,m}(t)$	Capacity allocated to user k at channel m
α, β, γ	Capacity allocated to RT, NRT and BE queues, respectively
I_{RT}, I_{ST}	PDR trend indicator for real-time and streaming traffic
$[H_{k,m}]^2(t)$	Channel gain of user k at sub-channel m at time t
$P_k(t)$	Priority metric of user k at time t
T_k	Minimum throughput requirement of streaming service of user k
$\lvert Q_k(t)\rvert$	Queue length of user k at time t
$P_{k,m}$	Power allocated user k at PRB m
r_k	Instantaneous throughput of user k
DB_{ST}	Delay budget of NRT streaming traffic
DB_{RT}	Delay budget of RT traffic
T^{NRT}	Throughput requirement of NRT traffic
$Y_{k,m}(t)$	Received signal at mobile user k on sub-channel m
$G_{k,m}(t)$	Complex Gaussian noise at user k on sub-channel m
N_{clus}	Number of clusters
μ_n	Number of centroids

3.3 Queue Sorting Algorithms in Traffic Differentiator

Two novel service-specific queue-sorting algorithms (SSSA) are proposed to prioritise users in RT and NRT queues, respectively in the *Traffic Differentiator* stage. This section describes the principle of these algorithms and the performance metrics used to evaluate the performance of these two algorithms.

Figure 3-4 shows the detailed functionality of *Traffic Differentiator* which mix traffic is differentiated into RT voice users, NRT streaming users, background or BE users and their control information is put in a separate queue. These four categories resemble four queues in Figure 3-1. Users carry their data packets in their individual queues, as shown in Figure 3-4.

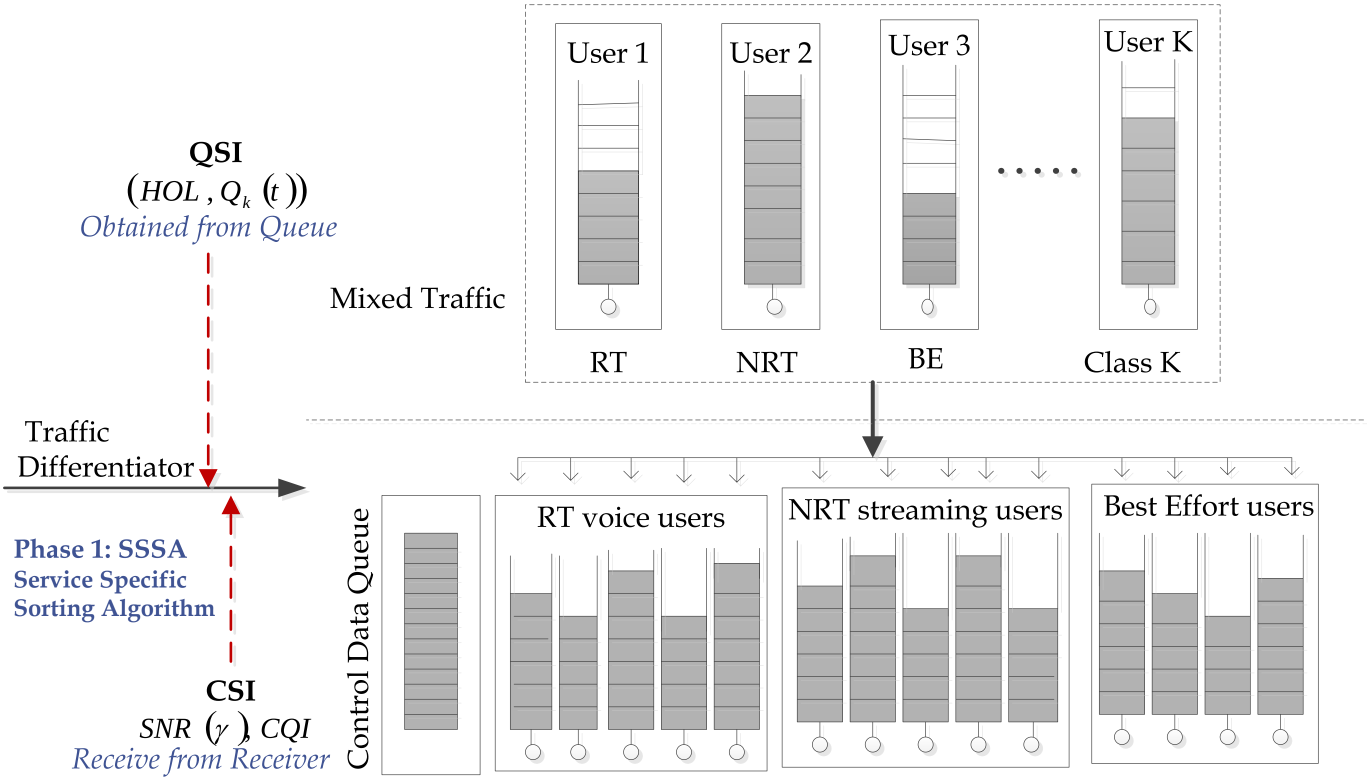


Figure 3-4 Traffic differentiator Stage

In each service queue, users require the same type of traffic e.g. Control queue contains only control information of all users, users in RT queue require voice service, users in NRT queue require streaming service and in BE queue users require background service. Users in these queues are sorted from the highest scheduling priority to lowest scheduling priority by applying SSSA, which uses QSI, QoS and CQI information.

The detailed description of queue sorting algorithms is given below.

3.3.1 FCFS Algorithm for Control Traffic Queue

Control information is transmitted on Broadcast Control Channel (BCCH) which is a downlink channel for broadcasting system control information [HAJMSIM06]. The control information consists of three types of information.

- scheduling information for DL data transmission,
- scheduling grant for UL transmission and
- ACK/NAK in response to UL transmission.

The control information needs to have the highest scheduling priority as it contains essential information to facilitate the actual resource allocation. Hence the control information is put into a dedicated queue and scheduled before all other service queues. As the control information of all users is equally important therefore it is proposed to transmit it in a FCFS manner. Physical Control Format Indicator Channel (PCFICH) dynamically indicates how many OFDMA symbols are reserved for transmitting control information. This can vary from 1 and 3 for each 1 ms sub-frame as shown in Figure 3-5 [HT09].

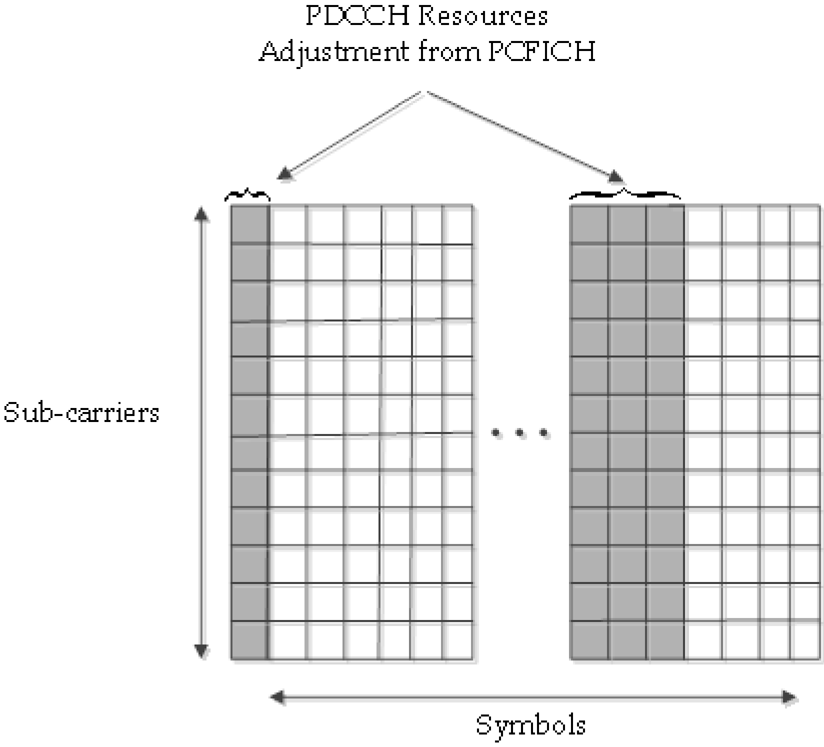


Figure 3-5 PDCCH resource allocation from PCFICH

Figure 3-5 shows the resource adjustment of PDCCH indicated by PCFICH. This thesis considers scheduling for DL data transmission, the control queue contains only scheduling information for the DL data transmission. And it is assumed that 1 OFDMA symbol per sub-frame is reserved for control information in each TTI for each user, which is enough for DL scheduling information. Thus, 1 OFDMA symbol/user is always reserved to transmit control data for the DL scheduling of each scheduled user in each TTI. For the resource allocation to remaining queues (RT, NRT and BE), the available PRBs are considered as the total number of PRBs subtracted by number of PRBs reserved for the control queue.

In this way, the control queue is always prioritised over other queues and is always allocated enough resources to transmit control information of the scheduled users.

3.3.2 Novel Queue Sorting Algorithm for RT Queue

The novel queue-sorting algorithm for RT queue considers the QoS requirements of real-time service and sorts users in this queue accordingly. In mobile networks, packet delay of users is considered an important parameter to evaluate the performance of real-time service. The packet delay experienced by RT users must be lesser than the delay upper bound, $d_k < DB_{RT}$ where d_k is the delay of HOL packet of user k and DB_{RT} is the upper bound or the delay budget for real-time service. HOL packet is the one ready to be scheduled i.e. at the front of user packet queue. If a user's HOL packet has delay more than delay upper bound , it is dropped from the queue otherwise it is scheduled. In addition to packet delay, this thesis considers two more performance indicators for real-time service; average PDR of real-time service and average PDR variation of RT users. These two parameters are discussed in detail in sections (3.4.1) and (3.4.2) respectively. The users in RT queue are sorted to give higher priority to the users with higher packet delays, longer waited queues and good channel conditions.

In the queue-sorting algorithm used for RT queue the priority metric is formed by the product of normalised HOL packet delay and the complex channel gain of the users, which is added to the square of the queue length of users as given in Equation 3-3. The normalised HOL delay is a ratio of user's HOL packet's waiting time and the delay budget for real-time service. The priority of user k at time t, $P_k(t)$ is,

$$P_k(t) = \left(F_k^{RT}(t) \times \left[H_k^{RT}(t)\right]^2\right) + [Q_k(t)]^2 \qquad (3\text{-}3)$$

In which $\left[H_k^{RT}(t)\right]^2$ is the channel gain of RT user k at time t, $Q_k(t)$ is the queue length of user k at time t and $F_k^{RT}(t)$ *is* normalised HOL delay of user k at time t, which is given by Equation 3-4.

$$F_k^{RT} = \frac{T_{waiting}^{RT}}{DB^{RT}} \tag{3-4}$$

In which $T_{waiting}^{RT}$ is the HOL packet delay of RT user k, which is very important to be considered because real-time service is highly delay-dependent. This parameter is updated in each TTI and it keeps packet delay under the delay budget. The rationale behind considering queue length of users is to further reduce the delay of RT users by allocating more resources to users with longer queues. Moreover the square of queue length is taken to increase the weight of queue length in the priority metric. This is because queue length is very important parameter to be considered in order to control the PDR of RT users. Also the priority metric considers the data rate of users by prioritising users with good channel conditions. It comes up with improvement in overall system spectral efficiency as well.

In the queue sorting process, users are sorted in descending order of this priority metric and users with higher values have higher priority.

The proposed delay-dependent queue sorting algorithm in Equation 3-3 leads to reduced packet delay of real-time service resulting in lower PDR.

This algorithm is able to control delay of RT users by appropriately setting priority level to different users.

To illustrate this principle, a system with two UEs demanding real-time service is shown in Figure 3-6.

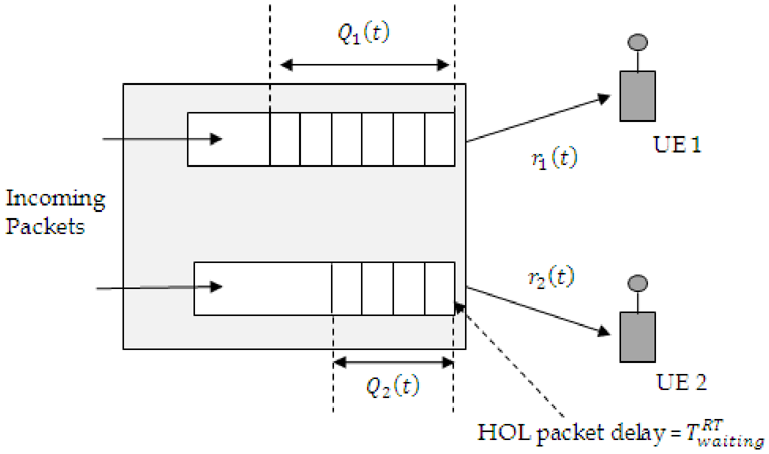


Figure 3-6 A system with two UEs

In Figure 3-6, $r_1(t)$ and $r_2(t)$ are instantaneous data rates and $Q_1(t)$ and $Q_2(t)$ are queue lengths of UE1 and UE2, respectively. As the channel conditions are instantaneously changing and the packets are generated by ON and OFF traffic model so instantaneous data rates and the queue lengths of both UEs are always changing. This property is used to make an efficient queue-sorting algorithm by prioritising users with higher HOL values, longer queues and good channel conditions, which results in improved QoS provision and good system spectral efficiency.

The parameter $F_k^{RT}(t)$ in Equation 3-3 embodies the QoS requirements and provides differentiation between UEs requiring real-time service. The objective is to keep the value of this parameter always lower than 1 to ensure that the packet delay of real-time service remains lower than the delay budget. To achieve this, in each TTI, the HOL packet delays for both UEs are calculated and the UE with higher HOL packet delay (not timed out) is prioritised over the other. In addition, queue length factor is used to get stable queues of both users by prioritising users with longer queue. In this way this algorithm exploits the variable channel and queue states and updates the priority of users accordingly to meet QoS demands of real-time service.

3.3.3 Novel Queue Sorting Algorithm for NRT Queue

NRT queue in the scheduling architecture consists of users requiring non-real time streaming video service. The novel queue-sorting algorithm for NRT queue sorts users in the queue according to the throughput requirement of streaming video service. The basic QoS requirement for streaming video service is the provision of long-term minimum throughput guarantee. Therefore QoS for streaming traffic is evaluated by the average achieved throughput of users of NRT queue. It must be equal to or greater than the minimum throughput requirements of streaming service, $r_k \geq T_k$, in which r_k is instantaneous throughput and T_k is minimum throughput requirement of streaming service. The minimum throughput requirement is taken as 240kb/s in this thesis as in [SYLXSYLX09]. To meet the requirement, this algorithm is designed to maximise the minimal throughput among all NRT users, which is given in Equation 3-5.

$$r_k = \max \left\{ \lim_{T\to\infty} \sum_{t=1}^{T} r_k(t) \geq T_k, k \in NRT\ Queue \right\} \tag{3-5}$$

The minimal throughput among all NRT users is given by Equation 3-6 as used in [SYLXSYLX09].

$$r_{min} = \min_{k \in NRT}(r_k) \tag{3-6}$$

To achieve this, the users in this queue are sorted to give priority to the users with low average achieved throughput.

The priority metric for streaming service is the product of the normalised HOL packet delay of each user, the ratio of its required throughput to the average throughput over a given time interval and the channel gain. The priority metric considers the product of these parameters to give equal weight to all parameters included in it. This is because these parameters equally contribute to the effective resource allocation to users belonging to the NRT queue.

The priority metric of user k at time t $P_k(t)$ is:

$$P_k(t) = F_k^{NRT}(t) \times E_k(t) \times \left[H_k^{NRT}(t)\right]^2 \tag{3-7}$$

In which, $F_k^{NRT}(t)$ is the normalised HOL packet delay of NRT user *k*, $E_k\,(t)$is the ratio of required throughput of streaming service to the average achieved throughput of NRT users and $\left[H_k^{NRT}(t)\right]^2$ is channel gain of NRT user *k*, at time *t*. The channel conditions of users are considered while prioritising users in the queue to improve the data rate of users thus improving overall system spectral efficiency.

The value of F_k^{NRT} is given in Equation 3-8 and the value of E_k is given in Equation 3-9.

$$F_k^{RT} = \frac{T_{waiting}^{NRT}}{DB^{NRT}} \tag{3-8}$$

In which DB^{NRT} is delay budget for streaming video service. To reduce the delay of streaming video packets, it is important to consider HOL packet delay of users belonging to this queue.

The parameter E_k embodies the throughput requirements of streaming video service. The value of E_k for NRT user k at time *t* can be calculated by Equation 3-9.

$$E_k(t) = \frac{T_k(t)}{R_k(t)} \qquad \therefore (T_{k(t)} = 240kbps) \tag{3-9}$$

In which $T_k(t)$ is the minimum required throughput of user k at time t, and $R_k(t)$ is its average achieved throughput which is calculated by (2-3). The rationale behind using this parameter is to keep each user's average achieved throughput equal or greater than the minimum throughput requirements of streaming service. As the main evaluation parameter for streaming service is minimum required throughput, it is very important to take into account of the minimum throughput requirements of it while prioritising users in this queue. To understand the logic of how this parameter leads to meet minimum throughput requirements, consider the following two cases.

Case 1:

In each TTI the value of E_k is calculated to evaluate the performance of streaming video service. Case 1 shows one scenario in which the average achieved throughput of user k is lower than its minimum throughput requirement.

$$E_k(t) = \frac{T_k(t)}{R_k(t)} > 1 \qquad (3\text{-}10)$$

Or, $$T_k(t) > R_k(t)$$

If the value of $E_k(t)$ is higher than one, it shows that the average achieved throughput of user k at time t is lower than the minimum throughput requirement of streaming video service. If this is the case the priority of user k is increased in the next TTI to meet the throughput requirements. This is achieved by considering the parameter E_k in queue-sorting algorithm for NRT queue. As given in Equation 3-7, the higher the value of E_k , the higher is priority of user k.

Case 2:

Case 2 shows that the average achieved throughput of user k is higher than or equal to its minimum required throughput.

$$E_k = \frac{T_k}{R_k} \leq 1 \tag{3-11}$$

Or, $$T_k \leq R_k$$

If the value of E_k is equal to or lower than one, it shows that the average achieved throughput of user k is equal to or higher than the minimum throughput requirement. In this case the priority of user k is either decreased ($if\ T_k < R_k$) or is kept same ($if\ R_k = T_k$).

In each TTI E_k is updated and users are prioritised accordingly, to meet throughput requirement of NRT users.

3.3.4 PF Algorithm for BE Queue

The BE queue in the proposed scheduling architecture represents the background service and does not have any QoS requirements. The priority is, thus, given to BE users based only on channel conditions. However to maintain fairness among users, the PF algorithm is used as the queue sorting algorithm for BE queue.

The priority of user k in BE queue at time t, $p_k(t)$ is:

$$P_k(t) = \frac{r_k(t)}{R_k(t)} \tag{3-12}$$

In which r_k is instantaneous throughput and R_k is the average throughput of user k at time t.

3.4 Machine Learning Algorithms in TD Scheduler

Packet scheduling is mainly focused on PRB allocation based on users' CSI, QCI information and QoS requirements. However because of ever increasing number of mobile users and limited number of available PRBs, it is difficult to guarantee all on-going users' QoS during the same TTI. Here arises the need of a TD scheduling algorithm to make decisions adaptively whether to admit or delay scheduling request of users, before actually allocating the radio resource to users in FD scheduling in each TTI. A novel Adaptive Time Domain Scheduling Algorithm (ATDSA) is proposed in this thesis which selects a pool of users adaptively from the service queues based on their priorities and the available radio resources users. It uses average PDR value of real-time voice & non-real time streaming video services to select a pool of users from the user queues.

The average PDR is calculated by Equation 3-13.

$$PDR = \frac{1}{K}\sum_{k=1}^{K}\frac{n_k^{dropped}}{n_k^{total}} \tag{3-13}$$

In which $n_k^{dropped}$ is the total number of packets dropped for user k, n_k^{total} is total number of packets arrived for user k and K is total number of active users.

This novel ATDSA works in the second stage of CLPSA, *TD Scheduler* that incorporates Hebbian learning process and K-mean clustering algorithm. Hebbian learning process is applied for adaptively allocating just enough radio resources to different services based on the average PDR of real-time and streaming services. K-mean clustering algorithm utilises average PDR information of individual RT users to further arrange the priority of RT users selected in *Traffic Differentiator* stage.

Figure 3-7 shows the second stage of CLPSA with two main process; Hebbian learning and K-mean clustering. These processes are described in detail in sections (3.4.1) and (3.4.2).

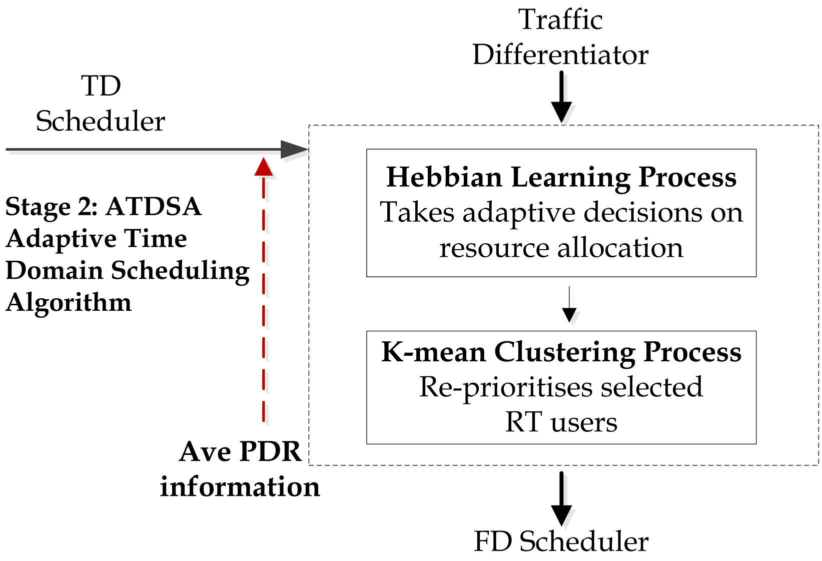

Traffic Differentiator
TD Scheduler
Stage 2: ATDSA Adaptive Time Domain Scheduling Algorithm
Hebbian Learning Process
Takes adaptive decisions on resource allocation
K-mean Clustering Process
Re-prioritises selected RT users
Ave PDR information
FD Scheduler

Figure 3-7 TD scheduler stage

3.4.1 Hebbian learning Process in ATDSA

Hebbian learning process takes decisions on radio resource allocation to real-time voice, non-real time streaming video and background services by adjusting the weight of these services during each TTI based on the average PDR values of the first two services. The aim is to allocate just enough resources to real-time voice and non-real time streaming video service, and assign the rest of radio resource to the background service.

In this process x, y and z are the weights of radio resource which should be allocated to real-time voice, non-real time streaming and background services, respectively such that,

$$x + y + z = 1 \tag{3-14}$$

The initial values of x, y and z are calculated based on the ratio of the number of active users in each of service type to the total number of active users in the system. It is given in Equation 3-15.

$$\begin{cases} x = \frac{X}{X+Y+Z} \\ y = \frac{Y}{X+Y+Z} \\ z = \frac{Z}{X+Y+Z} \end{cases} \tag{3-15}$$

In which, *X, Y* and *Z* represent the number of active users in real-time voice, non-real time streaming video and background service, respectively. The number of PRBs allocated to real-time voice, non-real time streaming and background services are represented by α, β and γ, respectively. The initial values of α, β and γ, are calculated by multiplying the rounded off values of x, y and z to the total number of available PRBs.

$$\begin{cases} \alpha = (round\ off\ x) \times M \\ \beta = (round\ off\ y) \times M \\ \gamma = (round\ off\ z) \times M \end{cases} \tag{3-16}$$

In which M is the total number of PRBs subtracted by the number of PRBs reserved for scheduling control data. In each TTI, the average PDRs values of real-time and streaming services are calculated and this information is saved in vectors *PDR_RT* and *PDR_ST,* respectively. The Hebbian learning process compares the average PDR value in the current TTI with PDR threshold, P_{TH} and the average PDR values in the previous TTIs and then based on it, makes resource allocation decision to real-time, non-real time and background services. According to 3GPP specification for LTE-A, P_{th} is 0.1.

However to avoid unnecessary resource allocation changes due to the time-variant feature of average PDR value, which is similar concept as to have a hysteresis in handover to avoid Ping-Pong effect, the average PDR trend indicators are used. These trend indicators indicate whether the average PDR is increasing by time or it is decreasing. For real-time traffic, the average PDR trend indicator is represented by I_{RT} and for streaming service it is represented by I_{ST}. These trend indicators keep a record of previous values of average PDR and allow resource allocation changes only after a consecutive number of average PDR increases or consecutive number of average PDR decreases.

To meet QoS requirements of the real-time service, the average PDR in the current TTI is compared with P_{TH} for real-time and average PDR in the previous TTI. The results derived from these comparisons for real-time service are saved in temporary parameters α_{RT}, β_{RT} and γ_{RT}. Similarly to meet QoS requirements of streaming service, the average PDR in the current TTI is compared with P_{TH} for non-real time and average PDR in the previous TTI. The results derived from these comparisons are saved in temporary parameters α_{ST}, β_{ST} and γ_{ST}. These temporary values are then used to make final decision on the values of α, β and γ in the next TTI.

The systematic flow of Hebbian learning process is shown in Figure 3-8.

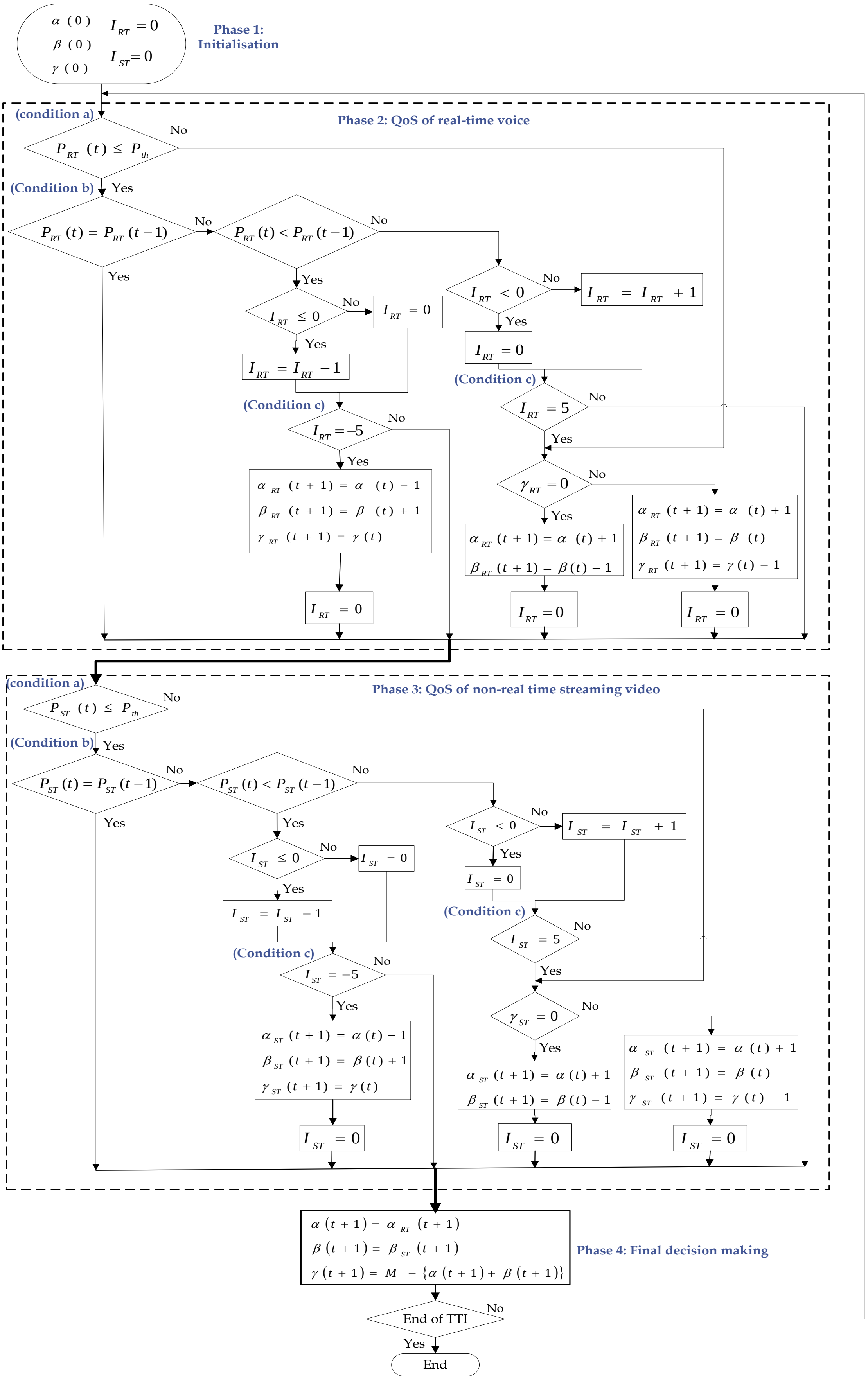

$\alpha(0)$
$\beta(0)$
$\gamma(0)$
$I_{RT}=0$
$I_{ST}=0$
Phase 1: Initialisation
(condition a)
Phase 2: QoS of real-time voice
$P_{RT}(t) \leq P_{th}$
No
(Condition b)
Yes
$P_{RT}(t) = P_{RT}(t-1)$
No
$P_{RT}(t) < P_{RT}(t-1)$
No
Yes
Yes
$I_{RT} < 0$
No
$I_{RT} = I_{RT} + 1$
Yes
$I_{RT} \leq 0$
No
$I_{RT} = 0$
Yes
$I_{RT} = I_{RT} - 1$
$I_{RT} = 0$
(Condition c)
$I_{RT} = 5$
No
(Condition c)
$I_{RT} = -5$
No
Yes
Yes
$\alpha_{RT}(t+1) = \alpha(t) - 1$
$\beta_{RT}(t+1) = \beta(t) + 1$
$\gamma_{RT}(t+1) = \gamma(t)$
$\gamma_{RT} = 0$
No
Yes
$\alpha_{RT}(t+1) = \alpha(t) + 1$
$\beta_{RT}(t+1) = \beta(t)$
$\gamma_{RT}(t+1) = \gamma(t) - 1$
$\alpha_{RT}(t+1) = \alpha(t) + 1$
$\beta_{RT}(t+1) = \beta(t) - 1$
$I_{RT} = 0$
$I_{RT} = 0$
$I_{RT} = 0$
(condition a)
Phase 3: QoS of non-real time streaming video
$P_{ST}(t) \leq P_{th}$
No
(Condition b)
Yes
$P_{ST}(t) = P_{ST}(t-1)$
No
$P_{ST}(t) < P_{ST}(t-1)$
No
Yes
Yes
$I_{ST} < 0$
No
$I_{ST} = I_{ST} + 1$
Yes
$I_{ST} \leq 0$
No
$I_{ST} = 0$
$I_{ST} = 0$
Yes
$I_{ST} = I_{ST} - 1$
(Condition c)
$I_{ST} = 5$
No
(Condition c)
$I_{ST} = -5$
No
Yes
Yes
$\alpha_{ST}(t+1) = \alpha(t) - 1$
$\beta_{ST}(t+1) = \beta(t) + 1$
$\gamma_{ST}(t+1) = \gamma(t)$
$\gamma_{ST} = 0$
No
Yes
$\alpha_{ST}(t+1) = \alpha(t) + 1$
$\beta_{ST}(t+1) = \beta(t)$
$\gamma_{ST}(t+1) = \gamma(t) - 1$
$\alpha_{ST}(t+1) = \alpha(t) + 1$
$\beta_{ST}(t+1) = \beta(t) - 1$
$I_{ST} = 0$
$I_{ST} = 0$
$I_{ST} = 0$
$\alpha(t+1) = \alpha_{RT}(t+1)$
$\beta(t+1) = \beta_{ST}(t+1)$
$\gamma(t+1) = M - \{\alpha(t+1) + \beta(t+1)\}$
Phase 4: Final decision making
End of TTI
No
Yes
End

Figure 3-8 Hebbian learning process

The functionality of Hebbian learning process passes through four phases, to reach the final decision. First phase is the initialisation phase, second phase is to investigate the QoS of real-time voice service, third phase is to investigate the QoS of non-real time streaming video service and the fourth phase is to make final decision on resource allocation to real-time voice, non-real time streaming video and background services. These phases with the detailed process involved in them, are described as below.

Phase 1*:* Initialisation

The learning process is initialised by the values of α, β, γ, I_{RT} and I_{ST}. In which α, β and γ, as mentioned earlier, represent the number of PRBs allocated to the real-time voice, non-real time streaming video and background services, respectively. These values are defined by Equation 3-16. To start with, the initial values of temporary parameters α_{RT}, β_{RT}, $\gamma_{RT,}$ α_{ST}, β_{ST} and γ_{ST} are taken equal to zero. And the trend indicators I_{RT} and I_{ST} are also initialised by zero value.

Phase 2*:* QoS of real-time voice

This phase takes into account of the comparison of average PDR value in the current TTI with the average PDR values in the previous TTIs and the P_{TH} for real-time service and saves results in α_{RT}, β_{RT} and γ_{RT}. Phase 2 includes the following conditions.

Condition (a): Compare current average PDR value with the PDR threshold

i. If the average PDR value in the current TTI is higher than P_{TH} for real-time service, the change in resource allocation to real-time service are triggered. And to increase resource allocation to real-time service, resource allocation either to streaming or background service is decreased. As streaming service has its own QoS requirements and is prioritised over background service. Therefore, first the value of γ_{RT} is checked. If $\gamma_{RT} > 0$, the value of β_{RT} is left unchanged, and the value of γ_{RT} is decreased by 1 PRB to increase the value of α_{RT} by 1 PRB. However if resource allocation to background service is zero i.e. $\gamma_{RT} = 0$ the value of β_{RT} is decreased by 1 PRB to increase the value of α_{RT} by 1 PRB. There is a change of 1PRB at each trigger because 1 PRB is a combination of 180 kHz (12 sub-carriers) and 1 ms TTI (14 OFDMA symbols), which is quite large chunk of spectrum to meet the requirement.

ii. If average PDR of real-time traffic in the current TTI is lower than or equal to P_{TH} , the average PDR values in the current and previous TTI are compared. And if average PDR values in the current and previous TTI are equal, the values of α_{RT}, β_{RT} and γ_{RT} are kept unchanged because the required QoS is already met for real-time service.

Condition (b): Compare current average PDR value with previous PDR value

i. If the average PDR value in the current TTI is lower than the average PDR value in previous TTI, the value of the trend indicator I_{RT} is checked. If the value of $I_{RT} \leq 0$, it is decreased by 1 indicating that average PDR of real-time service is decreasing with time. And if the value of $I_{RT} > 0$, its value is reset to zero so that it can be ready to count number of decrements in the value of average PDR. Reset is important to track consecutive decrements in the value of I_{RT}.

ii. If average PDR of the real-time service in the current TTI is higher than the previous TTI but lower than P_{TH} , the value of I_{RT} is checked. If $I_{RT} < 0$, it is reset it to zero. And if $I_{RT} > 0$ its value is incremented by 1, which indicates that the average PDR is increasing with time. Reset is important to track consecutive increments in the value of I_{RT}.

iii. If the average PDR value in current TTI is equal to the average PDR value in the previous TTI, no change either in α_{RT}, β_{RT} and γ_{RT} or I_{RT} is required as resource allocation to real-time service is sufficient.

Condition (c)*: *Avoid Ping-Pong effect

To avoid the Ping-Pong effect, real-time trend indicator is introduced which is initialised by zero value. It is incremented if the value of average PDR in the current TTI is higher than the value in previous TTI and vice versa. It only allows a change in the resource allocation to real-time service if its value reaches to +5 (in case of increments) or -5 (in case of decrements). This is to avoid too many changes in resource allocation due to time-variant changes in the value of average PDR. The indicator value is chosen ±5 because higher values may further improve the scheduling performance of real-time service in terms of reduced, average packet delay, average PDR, and PDR variation but it affects badly the performance of non-real time streaming video service in terms of achievable throughput. It also affects the system performance in terms of fairness among users and system throughput, by allocating more than enough radio resource to real-time service. To allocate just enough resources to real-time voice service, 5 is the most suitable value of the trend indicator.

i. If the value of I_{RT} becomes equal to -5, resource allocation α_{RT} to real-time service is decreased by 1 PRB, the resource allocation to streaming of β_{RT} is increased by 1 PRB. This is because streaming service is prioritised over background service. The value of I_{RT} is reset to zero. Reset is important to track consecutive increments in the value of I_{RT}.

ii. If I_{RT} becomes equal to +5, resource allocation to real-time is triggered. For this resource allocation to background service, γ_{RT} is checked. If $\gamma_{RT} = 0$, the value of β_{RT} is decreased by 1 PRB to increase the value of α_{RT} by 1 PRB. And if $\gamma_{RT} > 0$, the value of γ_{RT} is decreased by 1 PRB to increase the value of α_{RT} by 1 PRB by keeping the value of β_{RT} is unchanged. This is because streaming service is prioritised over background service due to its throughput requirements.

iii. Reset I_{RT} to zero. Reset is important to track consecutive increments in the value of I_{RT}.

Phase 3*:* QoS of non-real time streaming video

This phase takes into account of the comparison of the average PDR value of streaming video service in the current TTI with the average PDR values in the previous TTIs and the P_{TH} for non-real time streaming video service and saves results in α_{ST}, β_{ST} and γ_{ST}. The trend indicator is represented by I_{ST} in this phase. This phase repeats the condition from (a) to (c) (as in in phase 2) but for streaming video service.

Phase 4*:* Final decision making

Finally, a decision is made on the number of PRBs to be allocated to real-time voice service, non-real streaming video service and background service, in the next TTI,. These values are:

i. Resource allocation to real-time voice: α (t+1) = $\alpha_{RT}(t+1)$

ii. Resource allocation to non-real time streaming video: β (t+1) = $\beta_{ST}(t+1)$

iii. Resource allocation to background service: γ (t+1)=M-{α (t+1)+β (t+1)}

In which M is total number of available PRBs subtracted by the resource reserved for control data scheduling.

Hebbian learning process is repeated in each TTI.

Hebbian learning process thus allocates just enough resources to real-time voice and non-real time streaming video services and the rest to background service.

3.4.2 K-Mean Clustering Algorithm in ATDSA

An important criterion to evaluate fairness of real-time service is to measure the variation among average PDR values of RT users. The variation in average PDR is measured by taking the Standard Deviation (STD) of average PDR values of the group of all RT users, which is measured by the square root of the variance of average PDR values. The more the value of STD, the lesser the variance in average PDR values of RT users the higher is the level of average PDR fairness.

The STD of average PDR of RT users is given in Equation 3-18.

$$\sigma = \sqrt{\frac{1}{K}\sum_{k=1}^{K}(PDR_k - \mu)^2} \tag{3-18}$$

In which,

$$\mu = \frac{1}{K}\sum_{k=1}^{K} PDR_k \tag{3-19}$$

In which, σ is the variance (STD) in average PDR values of all RT users and μ is the mean value of average PDR values of all RT users.

The objective of using K-mean clustering algorithm is to minimise the variation in average PDR values among all RT users. Meanwhile it minimises the maximum value of PDR amongst all RT users.

The average PDR of user k is defined as the ratio of number of packets dropped to total number of packets arrived for user k as given in Equation 3-20 [SYLX09].

$$PDR_k(t) = \frac{n_k^{dropped}(t)}{n_k^{total}(t)} \tag{3-20}$$

To reduce variance and thus the maximum value of average PDR of the RT users, K-Mean clustering algorithm groups RT users based on their average PDR values. Users with higher and lower average PDR values are grouped separately and these groups are then sorted appropriately to give priority to users with higher average PDR values.

RT users are initially prioritised based on novel delay-dependent queue-sorting algorithm at the *Traffic Differentiator* stage. Delay-dependent queue-sorting only tries to reduce the packet delay (i.e. delay-aware) and does not specifically work for users with higher average PDR (i.e. PDR-blind). K-mean clustering however takes information on the average PDR values of individual users and its function is basically to move users with higher average PDR values to the front of queue thus increasing their priority. This is achieved by sorting clusters/groups (produced by K-mean algorithm) in the descending order of their mean value or centroid. However inside the clusters, priority order of users remains the same as at the *Traffic Differentiator* stage. In this process the number of clusters sets a trade-off between delay-dependent queue sorting and K-mean based priority order. Lower number of clusters leads to more weightage to the delay-dependent queue sorting as the priority of small number of users is changed. However higher number of clusters give more weightage to K-mean clustering priority as there is more granularity and in cluster-sorting process, prior priority of more users may change. For a good trade-off, two clusters are chosen for this work to keep both priorities meaningful and produce improved results in both domains i.e. packet delay and average PDR.

The schematic process of K-mean clustering algorithm is shown in Figure 3-9.

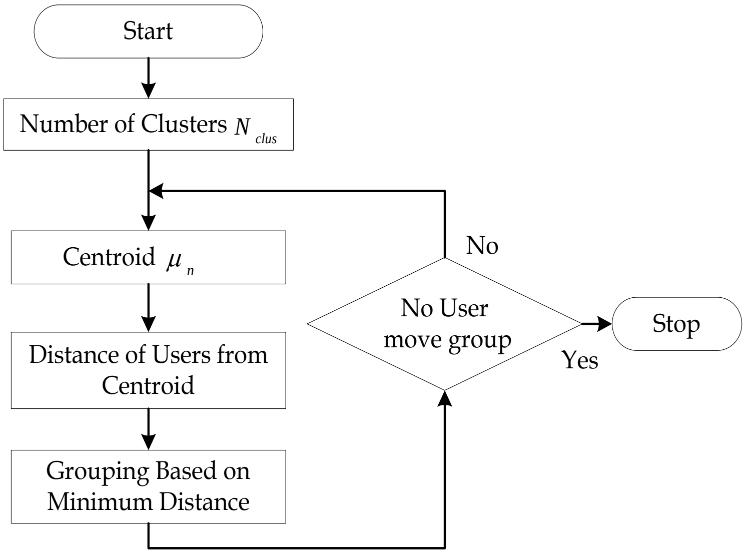

Start
Number of Clusters N_{clus}
Centroid μ_n
Distance of Users from Centroid
Grouping Based on Minimum Distance
No
No User move group
Yes
Stop

Figure 3-9 K-Mean clustering algorithm

Step 1: Average PDR of each user is calculated and saved in a vector S.

Step 2: Number of Clusters, N_{clus} are initialised with centroids $\mu_1, \mu_2, \dots \dots . \mu_n \in \mathbb{R}^n$ where centroid is similar to the centre of a cluster. In this thesis, two clusters are considered, so it defines two centroids.

Step 2: Each point of given data set S is assigned to its nearest Centroid based on Euclidian distance as given below.

$$N_{clus}^{i} = \min_{i=1,2,\dots.S} \|s_i - \mu_n\|^2 \qquad (3\text{-}21)$$

In which N_{clus}^{i} is the cluster to which s_i is assigned, μ_n is the n^{th} centroid and s_i is the i^{th} value in vector S.

Step 3: The mean of all points assigned to a Centroid is calculated and the position of each Centroid is updated by the mean of points assigned to it.

Step 4: Step 3 is repeated until no Centroid is shifted and no user associated with value *i* of set *S* move, resulting in 2 clusters.

Step 5: Repeat the process in each TTI.

3.5 Modified PF Algorithm in FD Scheduler

The selected users from each queue based on *Traffic Differentiator* and *TD scheduler* stage are allocated PRBs in the *FD scheduler* stage based on the values of α, β and γ, which are decided in the ATDSA. Figure 3-10 shows the Frequency Domain Resource Allocation (FD-RA) in *FD Scheduler* stage. It considers the list of prioritised users, checks per PRB CQI reports for each user and allocates PRBs to users by exploiting FD MU.

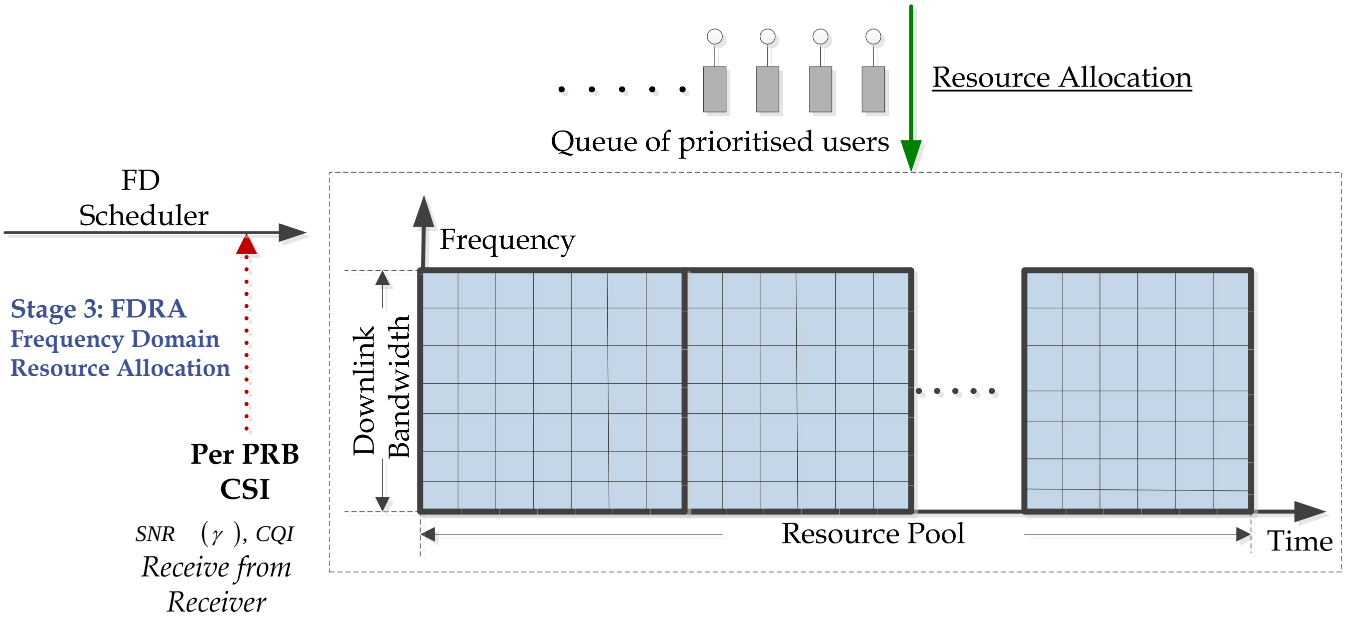


Figure 3-10 FD scheduler stage

The users are allocated PRBs in FD by using modified-PF algorithm. In the M-PF, the users are taken in their priority order. Channel conditions of the HOL user are calculated on each PRB relative to its average achieved throughput and the PRB on which the user shows the highest ratio is allocated to it. After this allocation, this PRB is deleted from the PRB list. Same process is repeated with all selected users until the available PRBs are finished. In this process the real-time voice, non-real time streaming video and background services are allocated α, β and γ number of PRBs, respectively.

Once a user is allocated a PRB, HOL packets from its buffer are deleted from its queue of packets and the queue is updated by Equation 3-22.

$$q_k(t) = Q_k(t) - \left[\frac{T_{len}}{s} \sum_{m=1}^{M} \rho_{k,m}(t) C_{k,m}(t)\right] \tag{3-22}$$

In which $q_k(t)$ length of user k at time t, $Q_k(t)$ is initial queue length of user k, T_{len} is length of time interval, s is packet size, M is total number of available PRBs and $\rho_{k,m}$ is 1 if the user is allocated otherwise it is 0. The queue $q_k(t)$ is updated after every TTI.

3.6 **System-Level Performance Indicators**

This thesis considers the system-level scheduling performance along with QoS requirements of users and uses the overall system throughput and user fairness to evaluate it.

System throughput at a given time t is calculated by the sum of average achieved throughput across all users including RT, NRT and BE users.

$$System\ throughput = \sum_{k=1}^{K} R_k(t) \tag{3-23}$$

In which $R_k(t)$ is the average achieved throughput of users k at time t.

System spectral efficiency is the ratio of system overall throughput and the total available bandwidth.

$$System\ spectral\ efficiency = \frac{\sum_{k=1}^{K} R_k(t)}{B} \tag{3-24}$$

In which B is the total available bandwidth of system.

Fairness in simple words means equal share of throughput among all users [CD07]. To measure the fairness among users, Raj Jain fairness index is adopted that is given in Equation 3-25 as used in [XC08] [CD07].

$$Fairness = \frac{\left[\sum_{i=1}^{K} R_k\right]^2}{K\sum_{i=1}^{K}(R_k)^2} \qquad (3\text{-}25)$$

The value of fairness index is 1 for the highest fairness when all users have same throughput. In Equation 3-25, K is the total number of users and R_k (updated by Equation 2-3) is the time average throughput of user k.

The PDR fairness is also considered in this thesis, which is discussed in section (3.4.2).

3.7 **Summary**

In this chapter the proposed cross layer design QoS packet scheduling architecture is described in detail. Novel algorithms used at different stages of this architecture are described. The novel SSSA is used at queue *Traffic Differentiator* stage to prioritise users in the RT and NRT queues. The novel ATDSA algorithm with Hebbian learning process and K-mean clustering algorithm is described at the *TD Scheduler* stage. Finally the frequency domain resource mapping is described at *FD Scheduler* stage. Next chapter describes the system-level simulation set up to validate the performance of proposed packet scheduling architecture. Some of the important simulation validation and verifications are also presented in chapter 4.

Chapter 4 System Level Platform

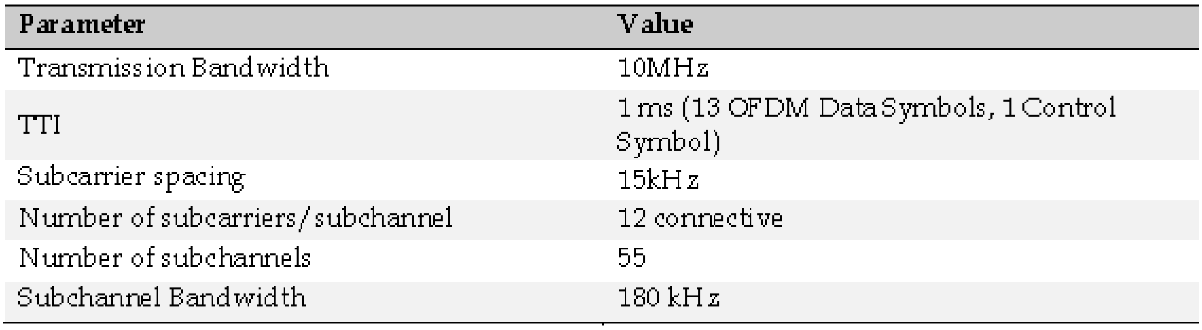
This chapter describes the simulation platform used to evaluate the performance of the proposed algorithms. It includes simulation model, simulation parameters, important simulation modules and the overall simulation flowchart. It also gives the wireless channel model used in this and simulation validation including validation of number of iterations.

4.1 Design of System Level Simulation Platform

The specifications of the system-level simulation are based on the LTE specifications defined by [3GPP09b]. The platform simulates an LTE-A networks that consists of a single cell with total system bandwidth of 10 MHz and PRB size of 180 kHz. The system bandwidth is divided into 55 PRBs and it works with a carrier frequency of 2 GHz. The wireless environment is the typical Urban Non Line-of-Sight (NLOS) and the path loss model is COST 231 Walfisch-Ikegami (WI) [Senarath07] which is used in many other LTE related publications [XC08] [XCSCZ11]. The delay budget for real-time service is 40 ms [SYLX09] [EFKMPTW06] and the required throughput for non-real time service is taken equal to 240 kbps as in [SYLX09]. The total eNB transmission power is 46dBm (40W) and the maximum Bit Error Rate (BER) requirement is 10^{-4} for all users. It is assumed as in [SYLX09] [JNTMM08] [GBP08] and [JL03] that the power allocation is uniform.

The summary of the downlink system parameters are listed in Table 4-1 [GBP08] [AZXY03].

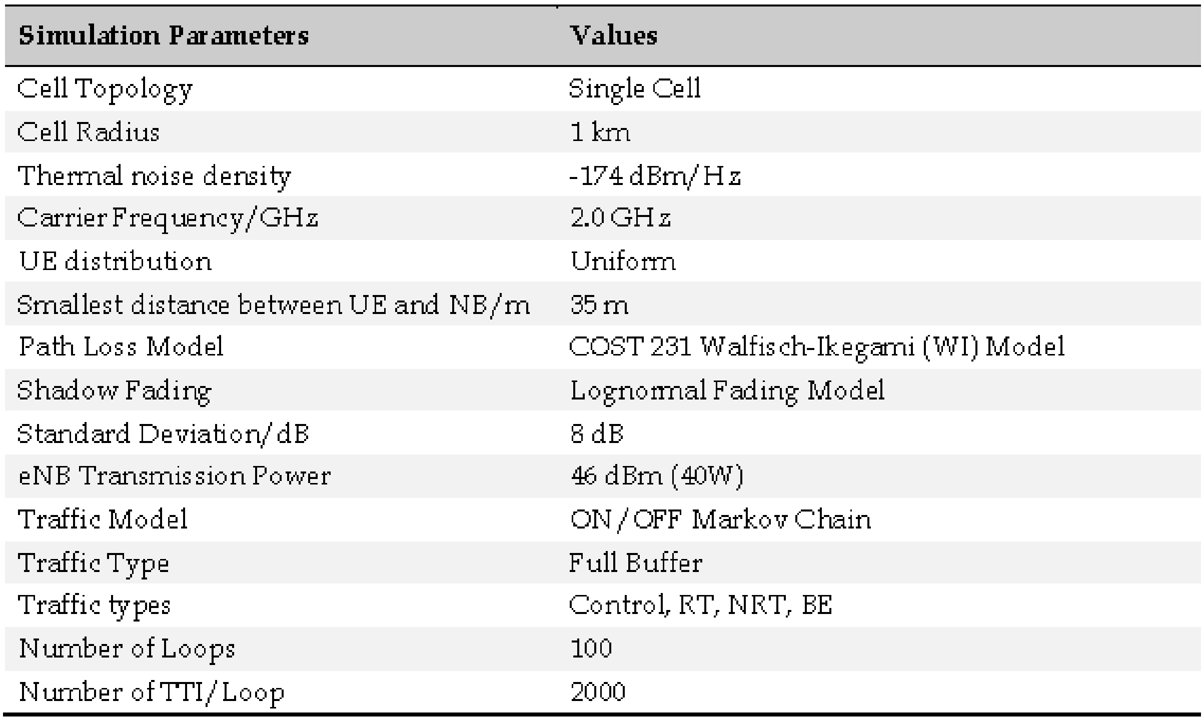
Table 4-1 Downlink transmission parameters



Parameter	Value
Transmission Bandwidth	10MHz
TTI	1 ms (13 OFDM Data Symbols, 1 Control Symbol)
Subcarrier spacing	15kHz
Number of subcarriers/subchannel	12 connective
Number of subchannels	55
Subchannel Bandwidth	180 kHz

The system-level simulation parameters are summarised in Table 4-2, which are the most commonly used in the literature on LTE-A.

Table 4-2 System-level simulation parameters



Simulation Parameters	Values
Cell Topology	Single Cell
Cell Radius	1 km
Thermal noise density	-174 dBm/Hz
Carrier Frequency/GHz	2.0 GHz
UE distribution	Uniform
Smallest distance between UE and NB/m	35 m
Path Loss Model	COST 231 Walfisch-Ikegami (WI) Model
Shadow Fading	Lognormal Fading Model
Standard Deviation/dB	8 dB
eNB Transmission Power	46 dBm (40W)
Traffic Model	ON/OFF Markov Chain
Traffic Type	Full Buffer
Traffic types	Control, RT, NRT, BE
Number of Loops	100
Number of TTI/Loop	2000

4.2 **Simulation Flow**

The simulation process is comprised of a number of simulation loops and each simulation loop consists of a number of TTI. In each TTI, a set of users is allocated radio resources based on the proposed SSSA and ATDSA in the TD, and M-PF algorithm in the FD. The main simulation modules are summarised below and their full descriptions are presented in section 4.2.

Initialisation module: It initialises the random positions of all UEs and the position of eNB. It computes the path loss and shadow fading of each user and creates the multipath fading in the cell.

Traffic module: It generates traffic for different services and keeps record of the number of packets generated, scheduled and dropped.

Traffic differentiating module: It divides Layer 2 (L2) buffer data into different service queues and sorts users within each queue.

CQI feedback module: It sends CQI feedback to different scheduling stages of the proposed packet scheduling architecture.

Average PDR information module: It calculates average PDR of real-time and non-real time service and also average PDR of individual RT users and sends this information to adaptive *TD scheduler* module.

Adaptive *TD scheduler* module: It collects the information on CQI and average PDR information from CQI feedback module and average PDR information module, respectively and then based on this information it takes decision on resource allocation to different services. It also re-arranges the priority of RT users based on their average PDR information on individual user-level.

Resource allocation module: It maps PRBs to the selected users based on resource allocation in adaptive *TD scheduler* module.

Channel capacity calculation module: It calculates the data rate of each user based on allocated PRB and CQI feedback.

System performance statistics module: It calculates system performance including QoS for different services, system throughput and user fairness.

The simulation uses time-stepping where each time step is equivalent to one TTI.

The overall simulation flow chart is shown in Figure 4-1, which includes the detailed processes involved. Final results are collected after the simulation has finished all TTIs and all simulation loops. The end of simulation loops indicates the end of whole simulation process.

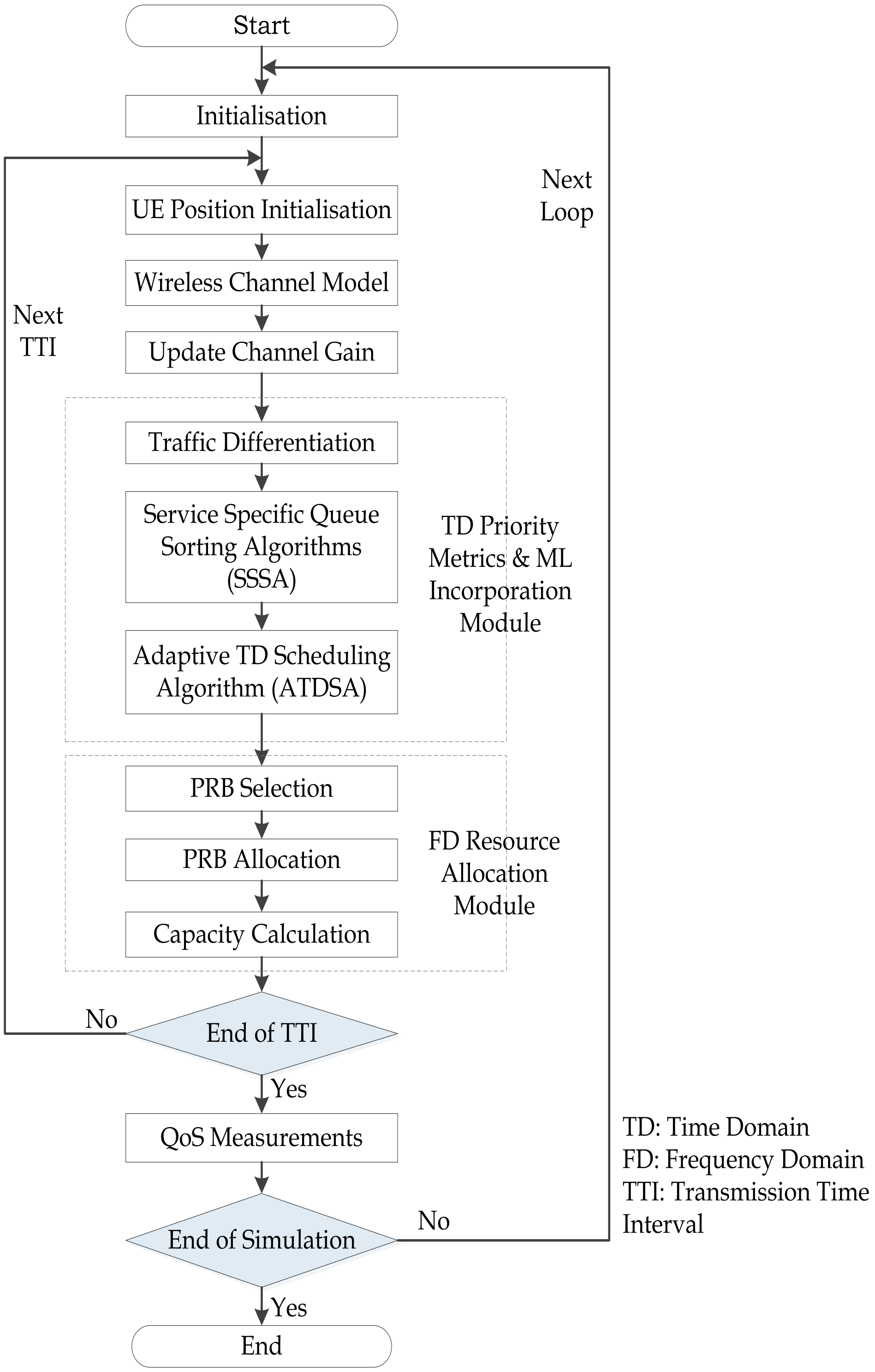


Figure 4-1 Flow chart of simulation platform

4.3 Module Functionality and Implementation

The functionalities of the main modules are described in the following.

4.3.1 Initialisation Module

This module initialises the system parameters and settings, and also initialises the positions of eNB and users. It creates shadow fading and multipath, and also calculates path loss. At the end of the module, it calculates the SNR of each user, which is used at different stages of packet scheduling architecture. The functionality flow of this module is shown in Figure 4-2.

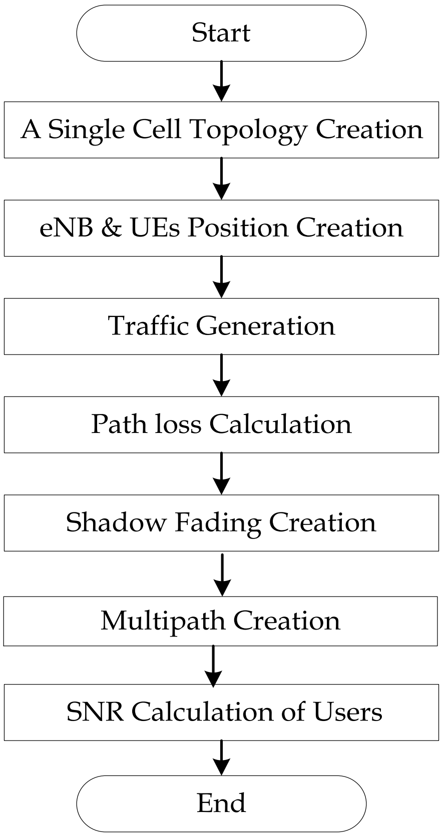


Figure 4-2 Flow chart of initial module

4.3.2 Traffic Differentiating Module

The main functionalities of this module are shown in Figure 4-3. It differentiates mixed traffic and generates service-specific queues. It listens to the feedback on SNR and the information on QoS parameters such as HOL packet delay, queue length and average achieved throughput, and then updates the priority of each user. In this work the CQI is referred to SNR of each user in each TTI. This information is then used in queue-sorting algorithms to sort the users of different services.

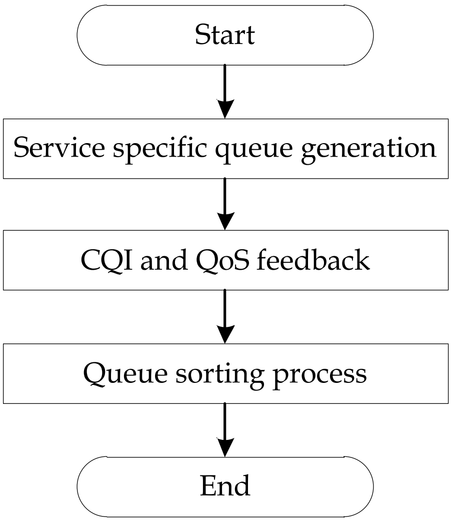


Figure 4-3 Flow chart of traffic differentiating module

4.3.3 Adaptive TD Scheduling Module

The functionalities of this module consist of two steps, which are shown in Figure 4-4 (a) and Figure 4-4 (b). The first step is to allocate the available radio resources among all services by using Hebbian learning process, while the second step is to further prioritise RT users based on their individual average PDR values by using K-mean clustering algorithm. The details of these two steps are described in sections 3.3.2.2 and 3.3.2.3, respectively.

Hebbian learning process selects a default policy at the initial distribution of PRBs among RT, NRT and BE queues. And then based on average PDR values of real-time and streaming services, it adaptively distributes PRBs to different services. The Hebbian learning process is repeated in every TTI for different channel environments followed by K-mean clustering algorithm.

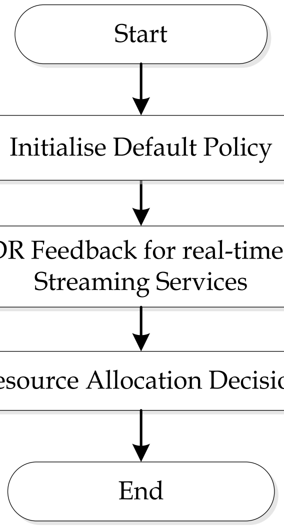

Start
Initialise Default Policy
PDR Feedback for real-time & Streaming Services
Resource Allocation Decision
End

Figure 4-4(a) Flow chart of Hebbian learning module

The systematic flow of second step is shown in Figure 4-4 (b). The K-mean clustering algorithm uses information of average PDR values of all RT users at individual user-level instead of average PDR of overall real-time service (as in 4-4 a). This information is used as an input vector for K-mean Clustering Algorithm to further prioritise RT user. Users with similar average PDR are grouped together and then these groups in the next TTI are sorted to give priority to the users with higher average PDR values.

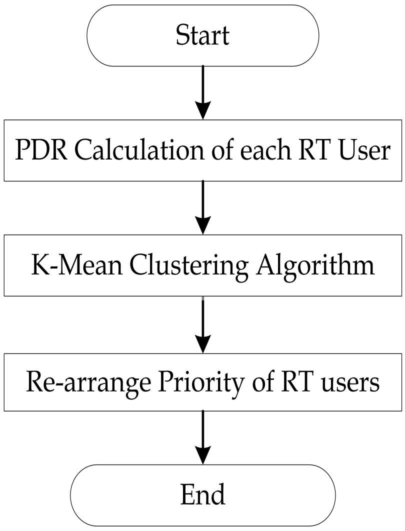

Start
PDR Calculation of each RT User
K-Mean Clustering Algorithm
Re-arrange Priority of RT users
End

Figure 4-4(b) Flow chart of K-mean clustering module

4.3.4 Resource Allocation Module

A systematic flow of resource allocation module is shown in Figure 4-5. The output of adaptive *TD scheduler* is a queue of users with set priorities. These users are allocated PRBs in the resource allocation module. This module picks users one-by-one based on their priority order and allocates PRBs to the users by exploiting FD MU diversity. After allocating PRBs, it calculates throughput of each user, system throughput and QoS provided to each user.

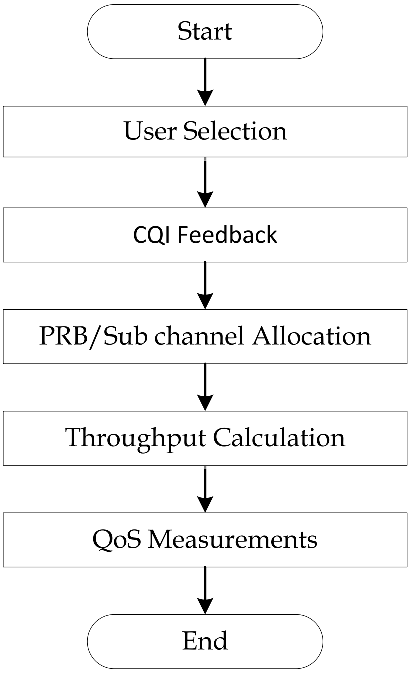


Figure 4-5 Flow chart of resource allocation module

4.3.5 Feedback Module

The flow process of feedback module is shown in Figure 4-6. This module calculates the user SNR in each PRB, throughput of all NRT users, HOL packet delay and average PDR of RT users. It then sends this information to *Traffic Differentiator* stage and adaptive *TD Scheduler* stage. The information on SNR, throughput and delay is sent to *Traffic Differentiator* stage; meanwhile the information of average PDR of real-time service and individual average PDR of RT users is sent to *TD Scheduler* stage. In addition, SNR information on each PRB will be also fed to the *FD Scheduler* stage.

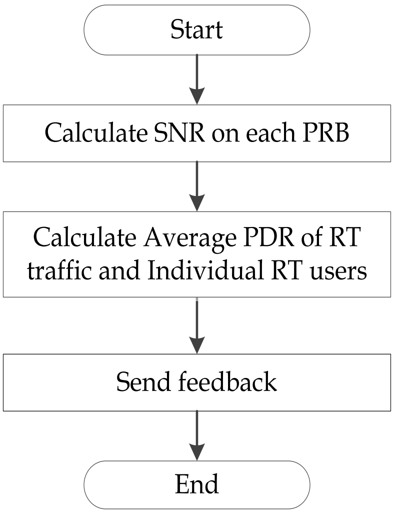

Start
Calculate SNR on each PRB
Calculate Average PDR of RT traffic and Individual RT users
Send feedback
End

Figure 4-6 Flow chart of feedback module

4.4 **Wireless Channel Model**

For the design of a radio communication system the information about radio communication channel characteristics is very important. Unlike fixed communication systems, the environment of wireless mobile communication systems is difficult to predict. Traditionally, radio channels are modelled in a statistical way using real propagation measurement data. Many measurements and studies have been done to obtain the characteristics of the radio environment. A good review of these measurements is given in [APY95] [FL96] [Mokdarl91] and [Hashmi93]. In general, signal fading in a radio environment can be divided into three components: a large-scale path loss, a medium-scale slow varying, and a small-scale fast fading. Accordingly, these three propagation components are used to describe a wireless environment that consists of path loss, shadowing (also called slow fading), and multipath (also called fast fading).

Figure 4-7 illustrates the three propagation phenomena occurring in wireless scenario.

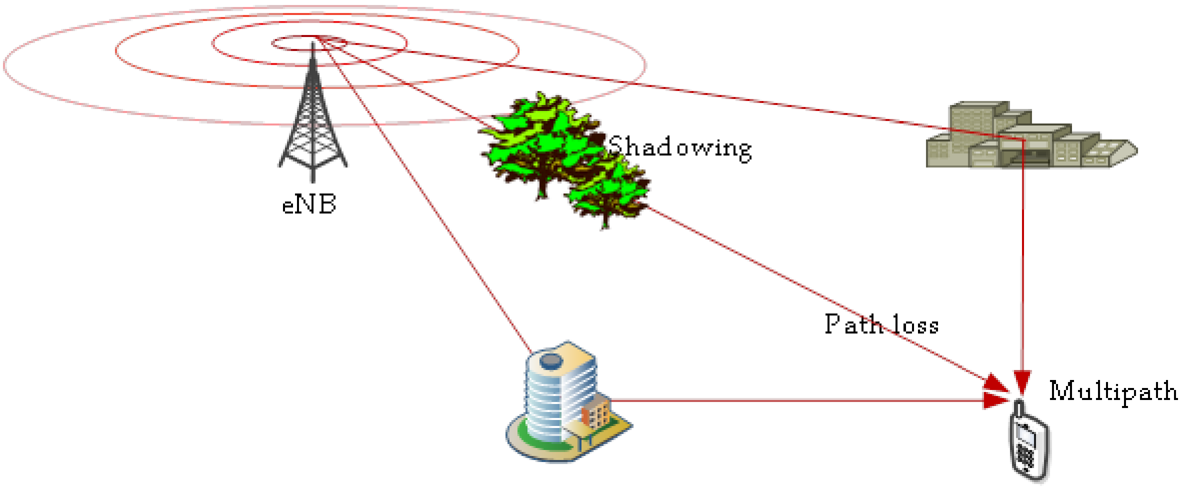


Figure 4-7 Radio channel attenuation

Path loss is phenomena of decreasing received signal power with distance due to reflection, diffraction around structures, and refraction with them. Shadowing is caused by obstruction of buildings, hills, trees, and foliage. Multipath fading is due to multipath reflection of a transmitted signal by objects like house, building, and other man-made structures, or natural objects such as forests surrounding a mobile unit.

In the large-scale transmission model path loss and shadow fading are influenced by the slow changes of average signal power and UEs positions. Meanwhile, in the small-scale transmission model multipath fading is mainly caused by the reflection, refraction and diffraction due to UEs movement and obstacles in the environment. It is used to describe fast change in the signal strength in short distances or short time.

The large-scale and small-scale changes both are considered in the simulation model. The characteristics of channel models are shown in Figure 4-8 [DPSB08].

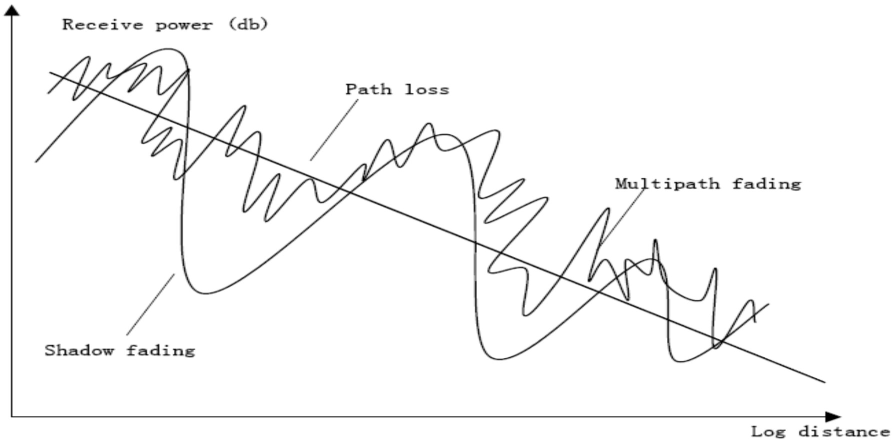


Figure 4-8 Transmission characteristics of a signal in wireless channel

Aforementioned the wireless environment in this simulation is considered NLOS in 2 GHz frequency range and COST 231 Walfisch-Ikegami (WI) model is used.

4.4.1 Path Loss Model

This model can be used for both urban and suburban environments. It is a combination of the models of J. Walfisch and F. Ikegami, which was enhanced in the COST 231 project; hence the name is COST 231 Walfisch-Ikegami (WI) model. Figure 4-9 [AWEweb12] shows the parameters considered in the computation.

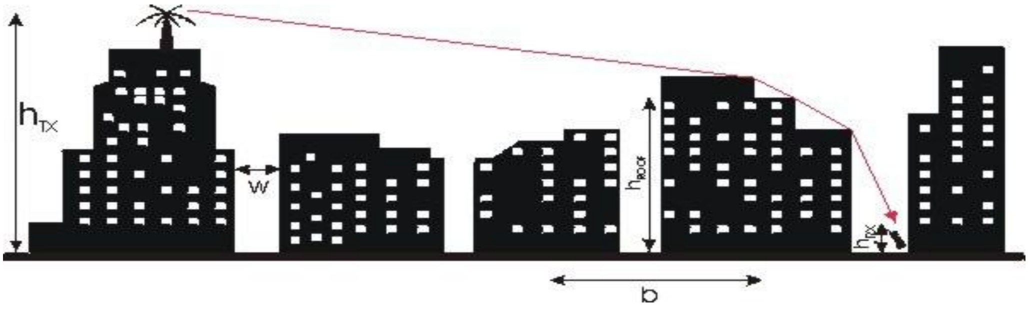


Figure 4-9 Parameters for computation

The model considers the buildings in the vertical plan between the transmitter (Tx) and the receive (Rx), in which street widths, building heights, transmitter and receiver heights are considered.

The accuracy of this model is quite high because in urban environments, the propagation in the vertical plan and over the rooftops i.e. multiple diffraction is dominating, especially if the transmitters are mounted above rooftop levels [AWEweb12].

The general parameters of the model are:

- Frequency (f): (800 to 2000MHz)
- Height of transmitter h_{T_x} : (4 to 50 m)
- Height of the receiver h_{R_x} : (1 to 3 m)
- Distance d between the transmitter and receiver: (20 to 5000 m)

The parameters depending on buildings are:

- Mean value of building heights: h_{roof}
- Mean value of width of streets: W
- Mean value of building separation: b

There are three terms which make up this path loss model [EHSB01].

$$L_b = L_0 + L_{rts} + L_{msd} \tag{4-1}$$

L_0= Free space loss

L_{rts} = Roof top to street diffraction

L_{msd}= Multi-screen loss

Free space loss is calculated as:

$$L_0 = 32.4 + 20\log(D) + 20\log(f) \tag{4-2}$$

In which, D is the distance of user from eNB measured in km and f is the carrier frequency measured in MHz.

Roof top to street diffraction L_{rts} is calculated as:

$$L_{rts} = -16.8 - 10log(w) + 10log(f) + 20log\left(h_{roof} - h_{mobile}\right) \qquad (4\text{-}3$$

In which,

$$\Delta h_{mobile} = h_{roof} - h_{mobile} \tag{4-4}$$

In which $h_{roof},$ h_{mobile} and Δh_{mobile} represent building height, Mobile Station (MS) height and the difference between building and MS heights measured in meters. And w is street width, which is also measured in meters.

The multi-screen loss L_{msd} is calculated as:

$$L_{msd} = L_{beh} + 54 + 18log(D) + K_f log(f) - 9log(b) \tag{4-5}$$

In which,

$$L_{beh} = -18log\left(1 + \left(h_{base} - h_{roof}\right)\right) \tag{4-5}$$

$$K_f = \left(-4 + 0.7 \times \left(\frac{f}{925} - 1\right)\right) \tag{4-6}$$

In which b is the building spacing measured in meters.

The value of b=60 m, w = 12 m, average height of roof is 25 m, average mobile height is 1.5 m and average eNB height is 40 m.

4.4.2 Shadow Fading Model

The correlation for shadow fading is defined as [3GPP09b]:

$$\text{Shadowing correlation}=\begin{cases} 0.5 & (cells) \\ 1.0 & (sectors) \end{cases} \tag{4-7}$$

$$\text{Shadow fading } SF_k = 10^{X_k/10} \tag{4-8}$$

In which, X_k is lognormal Gaussian distribution of user k such as:

$$X_k = aZ + bZ_i \text{ and } a^2 + b^2 = 1 \tag{4-9}$$

In which, Z and Z_k are two Gaussian random variables with an expectation 0 and standard deviation 8 dB. In addition, a and b are amplitudes of the real and imaginary parts of the signal, respectively. When $a^2 = b^2 = 0.5$, the correlation is 50% and when $a^2 = b^2 = 0$, the correlation is 100%.

4.4.3 Multipath Fading

Multipath fading or fast fading or multipath is an important character of wireless channel. When a radio signal is reflected or refracted in the transmission medium, they arrive at the receiver with more than one path. If the arrival time and the phase of the multiple signals are different then these signals can interfere constructively or destructively with each other at the receiver and the amplitude of resultant signal at the receiver changes dynamically. This scenario is called fast fading [TV05].

Rayleigh fading model is used in this simulation as it is suitable for NLOS urban environment. A complex number is used to express the amplitude and phase characteristics of the wireless channel. The real and imaginary parts of this complex number, MF=a + b*i are two Gaussian random variables with expectation 0 and standard deviation 1.

4.5 Radio Resource Allocation

Radio resource allocation in OFDMA-based LTE-A networks considers available spectrum resources both in time and frequency domains. Its aim is to effectively use spectrum resources and improve the network performance. The basic resource allocation unit in LTE-A network is PRB and one PRB is allocated to one user in the DL scheduling.

Let the total bandwidth be B, the total number of subcarriers be N, the bandwidth for each subcarrier be Δf which is equal to 15 kHz and one PRB is composed of 12 subcarriers. There are K users in the system and each user has data rate requirement R_k.

According to Shannon theory, the data rate is defined as a function of SNR and it can be expressed as,

$$R_k = \Delta f \times log_2(1 + \gamma_{k,m}) \tag{4-9}$$

In which, Δf is the PRB bandwidth, $\gamma_{k,m}$ is SNR of user k on RB m.

Based on Section 4.3, the wireless channel power gain for user k on subcarrier n experiences the path loss, shadowing fading and multipath fading. This can be expressed as,

$$[H_{k,n}]^2 = PL \times SF \times MF \tag{4-10}$$

For user k, let the power transmitted on subcarrier n be $P_{k,n}$ and σ^2 be the variance of AWGN on subcarrier n then the SNR for user k on this subcarrier is:

$$\gamma_{k,m} = \frac{P_{k,n} \times [H_{k,m}]^2}{\sigma^2} \tag{4-11}$$

Since the scheduling unit is PRB, let the transmission power on all PRBs be the same which is given as, $P_{k,m} = \frac{P_{total}}{M}$ where M is total number of PRBs, $P_{k,m}$ is the transmission power for user k on PRB m. Let $\gamma_{k,m}$ is the average SNR for user k on all the subcarriers in PRB m, $[H_{k,m}]^2$ is the channel power gain for user k on PRB m i.e. the average channel power gain for user k on all the subcarriers of PRB m. Then for user k, the data rate transmitted on PRB m is given as:

$$C_{k,m} = L \times \Delta f \times log_2\left(1 + \frac{P_{k,m} \times [H_{k,m}]^2}{\sigma^2}\right) \tag{4-12}$$

In which, L is number of consecutive subcarriers in a PRB.

4.6 **Simulation Validation**

The simulation model is an important groundwork of this thesis, so the simulation code is debugged line by line. Each module is run separately and then the simulator is analysed and validated by using classic scheduling algorithms such as RR, MAX C/I and PF in the *Traffic Differentiator* stage, and fair or strict pre-scheduling algorithms at the *TD Scheduler* stage. For this purpose three service queues are considered in which users are sorted by classic RR, MAX C/I and PF algorithms, respectively. In each simulation equal number of users and same network scenarios are taken into account to validate the results. The obtained results are same with theoretic results defined in the classic algorithms. Theoretically, as explained in Section 2.6, RR algorithm achieves the highest fairness level and the lowest system throughput. MAX C/I algorithm, on the other hand achieves the highest system throughput and the lowest fairness level. PF algorithm however makes a good trade-off between system throughput and fairness level by achieving system throughput higher than RR algorithm and fairness level higher than MAX C/I algorithm. The validation results consider system throughput achieved by RR, MAX C/I and PF algorithm.

4.6.1 Random Distribution of UEs

In the first step of simulation the system parameters and cell topology are initialised. The correlation between different network topologies is eliminated with random UE positions and wireless channel initialisation process. The wireless channel gain is updated in each TTI.

In the simulation single cell topologies are used and the performance of all packet scheduling algorithms is analysed under the same network conditions. Average results are taken by running the simulation over a number of iterations under a particular scenario. The radius of the cell is 1 km and all UEs are randomly distributed within the cell with minimum distance of 35 m from eNB.

The position of eNB is (0, 0). The position of eNB and all UEs in one simulation topology is shown in Figure 4-10 which shows the user random distribution characteristic.

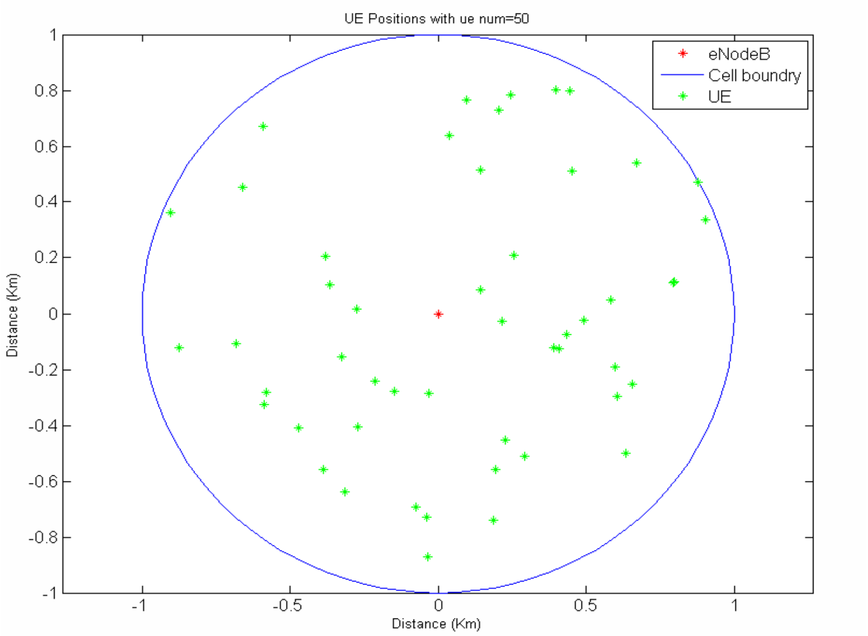


Figure 4-10 UE positions when UE number is 50

4.6.2 Verification of the Proposed Architecture

In this thesis, classic packet scheduling algorithms (PF, MAX C/I and RR) are simulated at the initial stage and they are used for verification of the proposed packet scheduling architecture. The behaviour of T*raffic Differentiator* and adaptive *TD scheduler* in the CLPSA is validated by classic PS algorithms and fair or strict priority scheduling, respectively. For this purpose, mix traffic is divided into three types representing three services; real-time data, non-real time data and background data. The queues are sorted by classic packet scheduling algorithms and users are picked from queues either by fair or strict priority scheduling. Then PRBs are allocated in the FD using 1 PRB per user strategy. The throughput of these services is calculated and compared to each other. The achievable data rate for user k on PRB m ($C_{k,m}$) is calculated by Equation (4-12). The expected results should show the highest throughput for MAX C/I algorithm, the lowest throughput for RR algorithm and PF algorithm should show a system throughput higher than RR and lower than MAX C/L algorithm. Furthermore due to TD and FD MU diversity, the system throughput for the algorithms increases as the number of users increases unless the system reaches its saturation point.

Figure 4-11 shows the average throughput achieved by RR, MAX C/I and PF algorithms when number of active users is 20. It shows the behaviour of *Traffic Differentiator* three separate simulations are run, where RR, MAX C/I and PF algorithms are used as queue sorting algorithm, respectively. In the first simulation users in all queues are ordered in RR manner, in the second simulation users in all queues are sorted by MAX C/I and in third simulation by PF algorithm. The results are then compared by showing the average achieved throughput by each simulation.

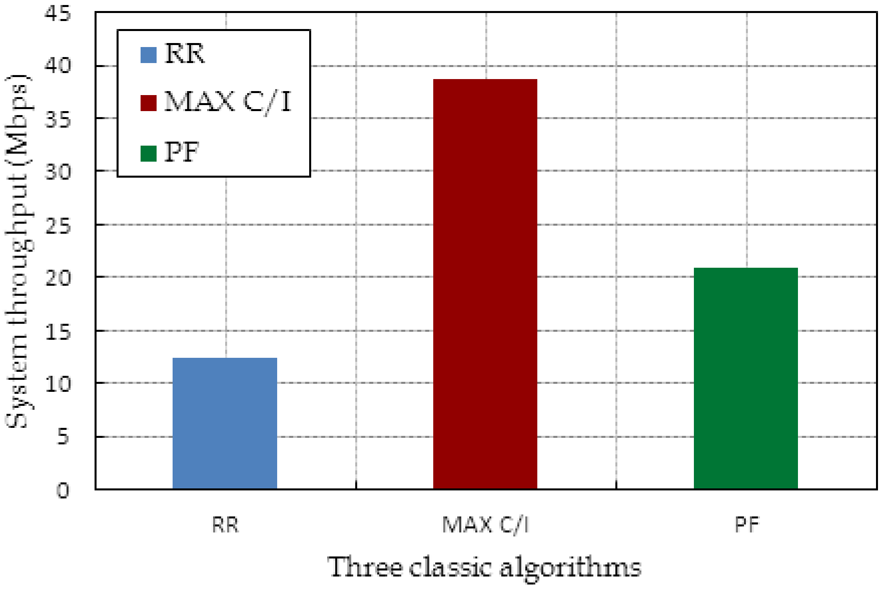


Figure 4-11 System throughput for 20 users

As expected, MAX C/I has the highest throughput as compared to RR and PF algorithms, RR algorithm has the lowest throughput and PF algorithm has the throughput in between the MAX C/I and RR algorithms. As shown in the Figure 4-11 the throughput for MAX C/I algorithm is 29 Mbps higher than RR algorithm and 10 Mbps higher than PF. This is because in MAX C/I algorithm, the users with the good channel conditions are scheduled.

The validation is continued by running one more simulation in which the number of users is varied and RR, MAX C/I and PF algorithms are used to sort users in the three queues representing real-time, non-real time and background services, respectively. As expected the average throughput increases as the number of users increases provided the availability of PRBs. It reaches at the saturation point when the number of users becomes equal to the number of available PRBs. Figure 4-12 shows average achieved throughput under different system load.

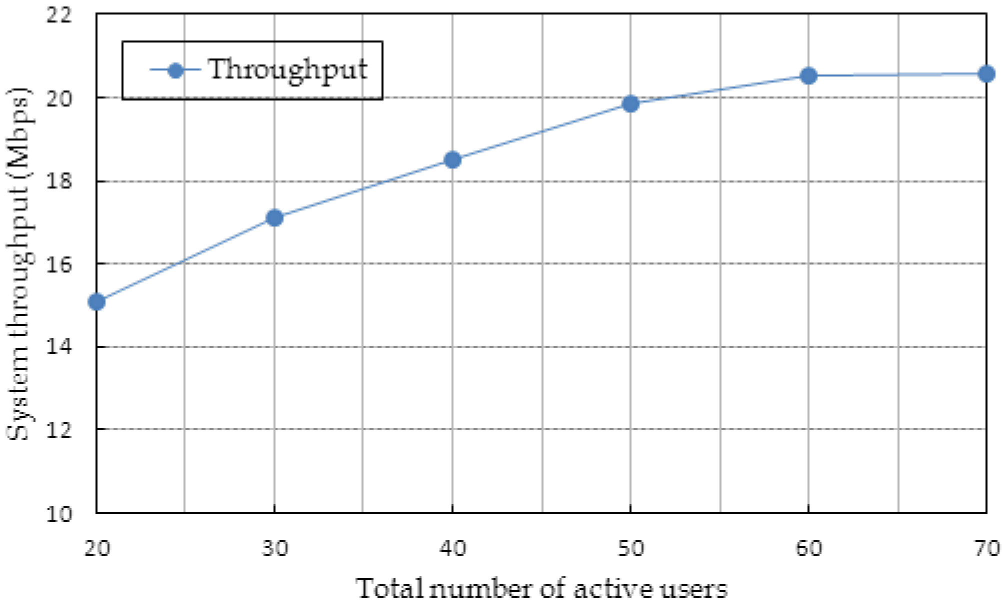


Figure 4-12 System throughput for different number of active users

It is shown that by increasing number of users, system throughput increases unless users are more than the number of PRBs and system reaches the saturation point.

The validation is also conducted by taking the throughput of users at TTI level so that instantaneous throughput of users can be analysed to validate the random changes in user positions and network conditions. The example of the instantaneous throughput of users along number of TTIs is shown in Figure 4-13.

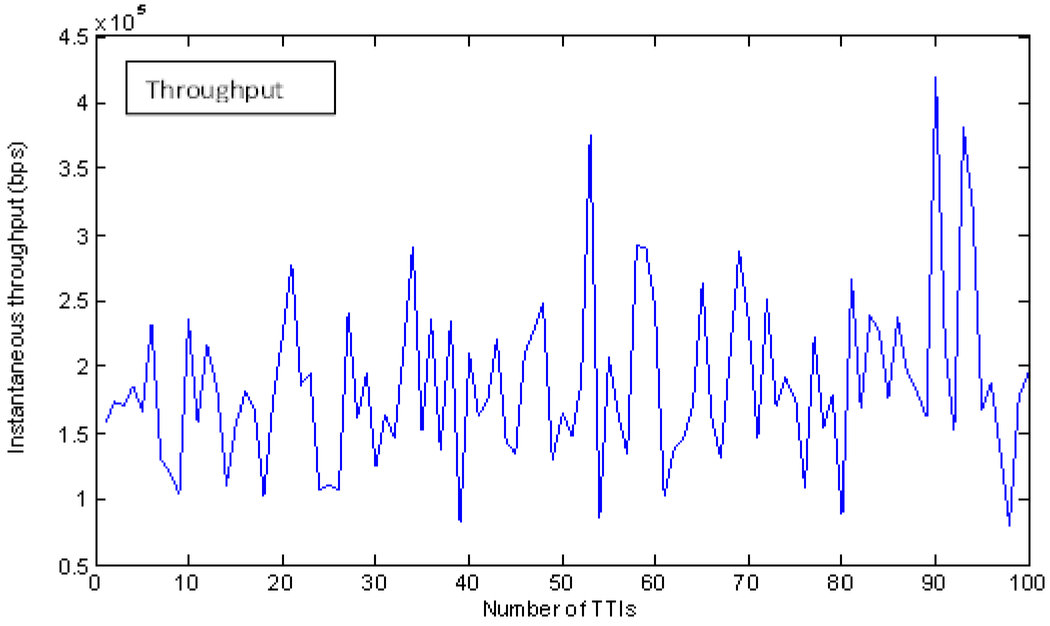


Figure 4-13 System instantaneous throughput

Figure 4-13 shows instantaneous throughput of 70 users at each TTI when 100 number of TTIs are used, which varies randomly due to time varying wireless channel conditions and changing positions of users. For system level throughout, instantaneous throughput on all TTIs is added and averaged over number of total active users.

4.6.3 Verification of the Number of Iterations

The number of simulation iterations impacts the simulation results a lot. This is because the locations of users will change randomly in each simulation loop and wireless channel conditions are time variant. To combat the effect of randomness and of time variant characteristics of channel conditions, large number of TTI is recommended, for proper simulation results.

From simulation point of view, the higher the number of simulation iteration, the longer the simulation time. Therefore a verification of deciding a reasonable number of iterations for simulations is required. Therefore average system throughput is calculated with different number of simulation iterations using the simulation parameters given in Table 4-1. The comparison of average system throughput with different number of simulation iterations is shown in Figure 4-14. In this comparison, there are 70 active users and the number of TTI is 2000 in each simulation. The simulation has been run for 100 times to collect 100 samples and the data is analysed and plotted as in Figure 4-14.

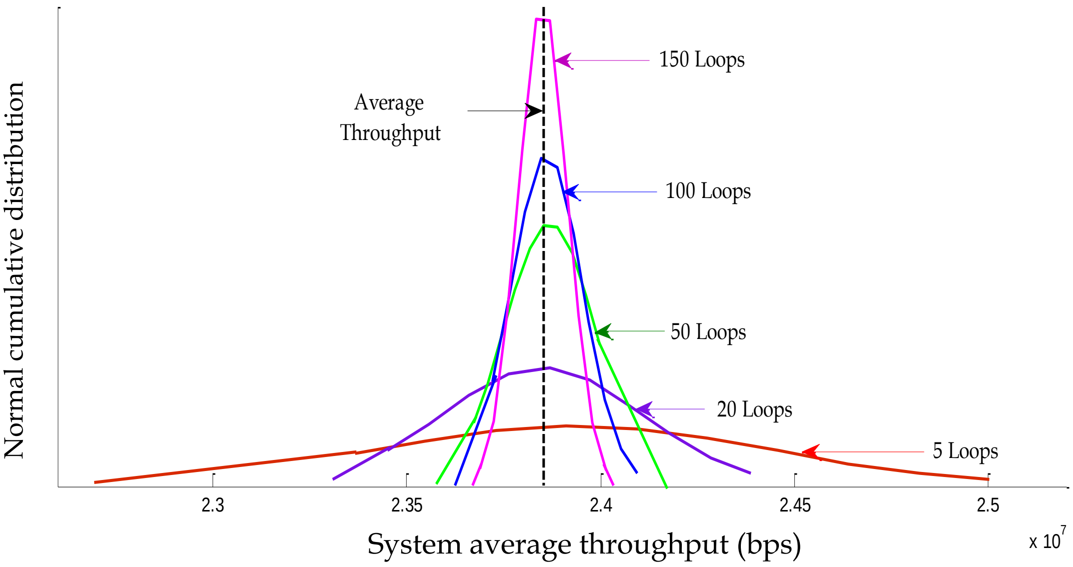


Figure 4-14 System average throughput vs. number of loops

The data is analysed for 5, 20, 50, 100 and 150 numbers of simulation iterations. The black dotted line in the centre is the average value from all samples and the curves in colours show the normal cumulative distribution curve from different data sets. The curves for 5, 20 and 50 number of simulation iterations show large standard deviation. However the curves for 100 and 150 loops show almost same standard deviation with 95% confidence interval, which is acceptable. Hence, in this thesis all the simulations are conducted with 100 numbers of simulation iterations.

4.7 Summary

In this chapter the system-level simulation set up is presented, which is used to analyse the performance of packet scheduling architecture. The overall flow chart and the detailed modules are described. Simulation parameters, wireless channel model and radio resource allocation model are also presented in this chapter. In addition, simulation validation and verification of number of iterations is presented. Simulation results on all algorithms including SSSA, Hebbian learning and K-mean and the joint performance of all algorithms are given in the next chapter.

Chapter 5 Simulation Results and Analysis

This chapter shows the simulation results and their analysis from different perspectives, to evaluate the performance of the proposed scheduling architecture. These results show the performance improvements of the proposed, SSSA, Hebbian learning process, K-mean clustering algorithms separately and the performance of overall scheduling architecture. The performance of overall scheduling architecture is tested under different network scenarios such as changing traffic split among real-time, streaming and background services and by varying system load, to exemplify the real network.

The results are compared against state-of-the-art QoS aware Mixed Traffic Packet Scheduling algorithm in UTRAN Long Term Evolution Downlink (abbreviated as MIX) [JNTMM08], Sum Waiting Based Scheduling (SWBS) [SYLX09] and classic PF algorithm. In MIX [JNTMM08], mix traffic is differentiated into service queue, conventional algorithms including PF and MAX C/I are used to sort users in the queues of different services and in the TD, fair scheduling methods are used to pick users from the queues. The second reference algorithm SWBS [SYLX09] uses the sum of waiting time of all packets of real-time and non-real time services, respectively to prioritise users in the queues. Waiting time for real-time service is the time during which packets remain in the queue to be scheduled and for non-real time service it is calculated by considering the throughput requirements of this service (section 2.6.4). Users maximising the product of sum waiting time and channel gain (Equation 2-7) are prioritised. While in the proposed architecture CLPSA, novel service-specific queue-sorting algorithms (SSSA) are used to prioritise users in RT and NRT queues, the conventional PF algorithm is used for BE queue and FCFS is used for the control data queue. In the TD, the scheduling architecture uses novel ATDSA to allocate radio resource efficiently to the prioritised users in RT, NRT and BE queues. In the *FD Scheduler* stage, PRBs are mapped to the selected users by M-PF algorithm.

5.1 **Traffic Model**

An ON-OFF source is used to generate packets for real-time voice service, streaming video service and background service. It uses only two states, i.e. ON and OFF states and the time spend between these two states is referred as transition time. The transition time is expected to follow an exponential distribution [Adas97]. For each user, the source generates packets at constant rate. The ON-OFF source is characterised by L, which is the mean number of packets generated during the ON time, the peak rate during ON time S and the mean rate of total ON and OFF time r. These factors determine the mean duration of the ON and OFF states. The equilibrium probability of the ON state of the source is calculated as, φ=*r/S*. As the ON and OFF states are exponentially distributed, the source can be modelled by two-state Markov chain. The mean packet generation rate during ON time is represented by L and is assumed to be much greater than 1, i.e. $L >> 1$. The transition probabilities are represented by x and y. The transition rates of the source from ON to OFF and vice versa, is calculated as follows [Chandrasekaran06]:

$$t_1 = (from\ the\ OFF\ to\ the\ ON\ state) \colon \varphi S/\big(L(1-y)\big) \tag{5-1}$$

As, $\varphi = r/S$, which basically means $(1 - x = r/S)$ therefore $S = r/(1 - x)$. Substituting value of "φ" in (5-1),

$$t_1 = \frac{r}{S} \times S \times \frac{1}{L(1-y)} \tag{5-2}$$

Or, $t_1 = r/L(1 - y)$ (5-3)

$$t_2 = (from\ the\ ON\ to\ the\ OFF\ state) \colon S/L \tag{5-4}$$

Substituting value of "S" in (5-4),

$$t_2 = \frac{r}{(1-x)} \times \frac{1}{L} \tag{5.5}$$

Or, $t_2 = r/L(1-x)$ (5.6)

Therefore,

$$t_1 = r/L(1-y) \tag{5-7}$$

$$t_2 = r/L(1-x) \tag{5-8}$$

The probability of mean "ON" time is 0.35 and the user remains active for 0.1 second in the "ON" time. The packets are generated at constant rate, which is 3 packets in every "ON" time for real-time service, 2 packets for non-real time streaming video service and 1 packet for background service [Chandrasekaran06].

The ON-OFF source is shown in Figure 5-1[Chandrasekaran06].

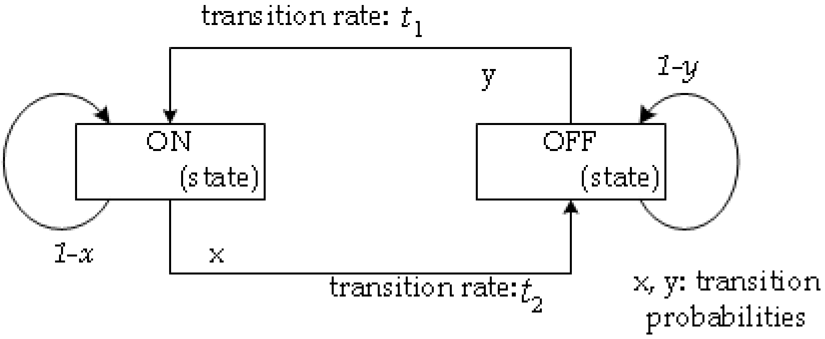


Figure 5-1 ON-OFF traffic model

5.2 User Queuing Model

The queuing model of a user is shown in Figure 5-2. Incoming packets are buffered in the queue until they are transmitted. The delay of packets increases as they wait in the queue [31]. The HOL packet has the longest delay as compared to the delay of all other packets in the queue.

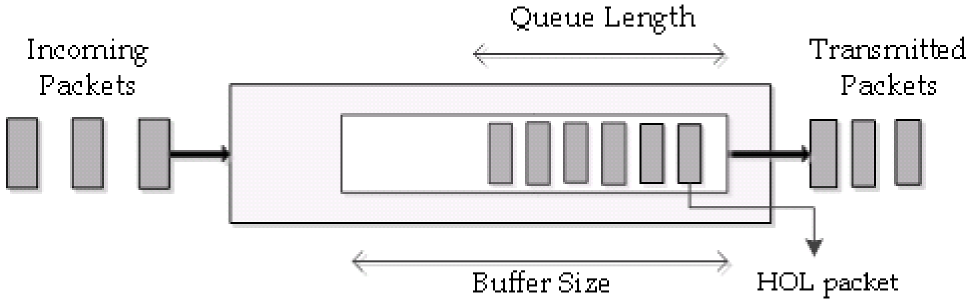


Figure 5-2 A user's queue model

When a user is scheduled, its HOL packets are transmitted first and the queue is updated by deleting the number of packets transmitted by using (Equation 3-21).

5.3 Performance Evaluation of SSSA

This section presents the effectiveness and performance improvement of novel service specific queue sorting algorithms, SSSA for real-time and streaming video services. The simulation results of SSSA show that *average delay*, *variation in average PDR of RT users* and *average PDR of real-time service* are reduced and *throughput requirement* of streaming video service is fulfilled while achieving good *system spectral efficiency*.

In this section, the proposed SSSA is compared with QoS aware SWBS algorithm to evaluate the user-level and the system level performance of the SSSA.

The system model of simulations in this section is shown in Figure 5-3.

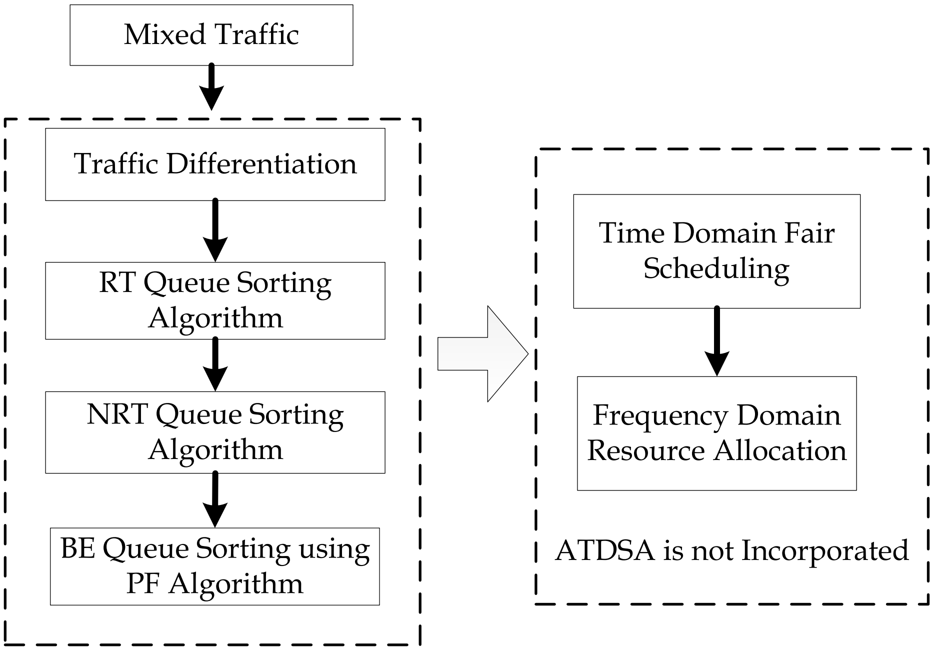

Mixed Traffic
Traffic Differentiation
RT Queue Sorting Algorithm
NRT Queue Sorting Algorithm
BE Queue Sorting using PF Algorithm
Time Domain Fair Scheduling
Frequency Domain Resource Allocation
ATDSA is not Incorporated

Figure 5-3 System model of SSSA

The input is mix traffic, which is differentiated into queues based on the requirements of users. Users in the RT queue and NRT queue are prioritised by the novel SSSA, users in the BE queue are prioritised by conventional PF algorithm and control data packets are transmitted in FCFS manner. The prioritised users are then picked one-by-one using fair scheduling in the TD and finally allocated PRBs in the FD using M-PF algorithm. Novel ATDSA, which is proposed in the TD to intelligently distribute the available PRBs among all services, is not implemented in these results. This is because this section presents the effectiveness of SSSA, separately.

5.3.1 QoS Performance of Real Time and Streaming Services

QoS of real-time service is evaluated by *average delay* and *average PDR* of real-time service and *average PDR variation* of RT users. QoS of non-real time service is evaluated by *average* *achieved throughput* of streaming video service. The *average achieved throughput* of background service is also presented in this section.

The number of RT users is taken equal to the total number of NRT and BE users representing equal load from real-time and non-real time services, as used in [SYLX09].

Figure 5-4 shows the average delay of real-time service for SSSA and SWBS algorithms with system load.

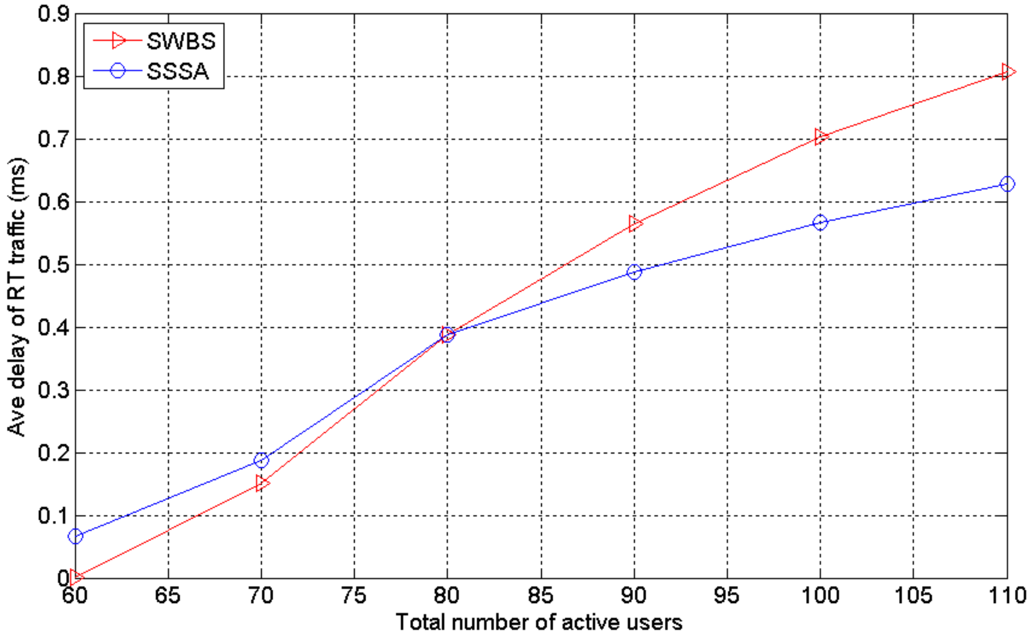


Figure 5-4 Average delay of real-time service

The proposed SSSA shows slightly higher delay than the reference SWBS algorithm up to a load of 80 active users however, which becomes lower than SWBS at system loads higher than 80 active users. This is because the proposed SSSA considers QSI in terms of queue length of users, to prioritise RT users. Queues do not build up at lower system loads, where packets of all users are scheduled. At higher system loads when competition to get radio resource is high, each user's queue becomes longer and longer. In this scenario, the consideration of queue length in queue-sorting becomes effective and lowers the average packet delay. The SWBS algorithm on the other hand, only considers HOL packet delay of users and is able to reduce average packet delay at lower system loads than 80 active users. However at lower system loads, packet delay shown by the proposed SSSA remains under the delay upper bound.

Figure 5-5 shows average PDR of real-time service under different system loads.

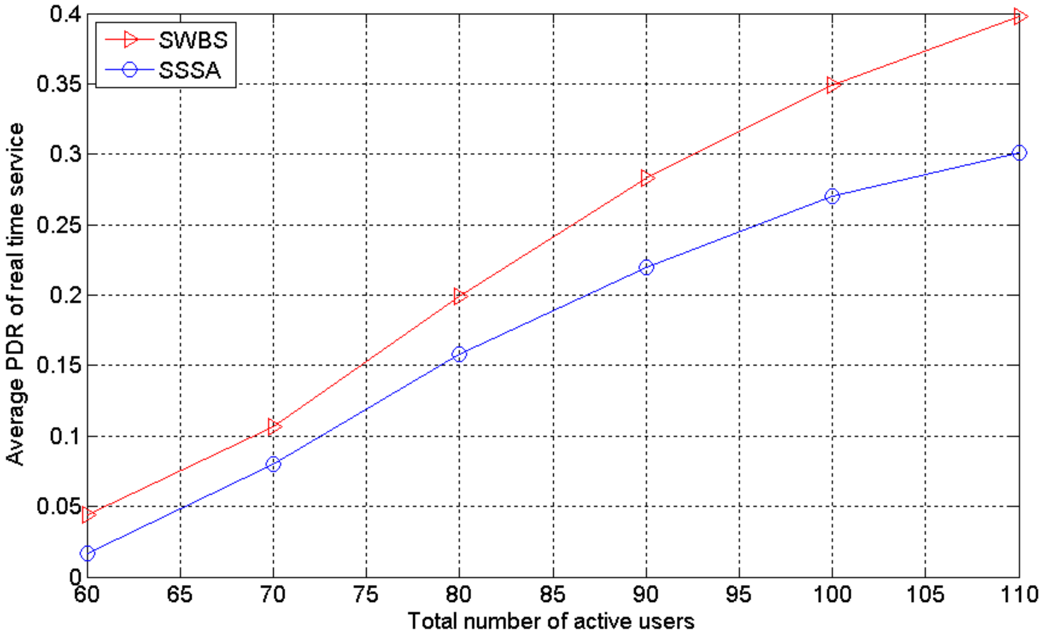


Figure 5-5 Packet drop rate of real-time service

The proposed SSSA has lower average PDR of real-time service than SWBS at all system loads. This difference becomes 10% in highly loaded conditions where the total number of active users becomes 110. This is because SSSA updates the priority of RT users based on both queue length and HOL packet delay information. It gives high priority to the users with higher packet delay and longer queues to reduce average delay of users. When users have lower delays, packets in their queues do not exceed their delay bounds leading to lower average PDR.

Figure 5-6 shows the variation in average PDR of all RT users under different network loads.

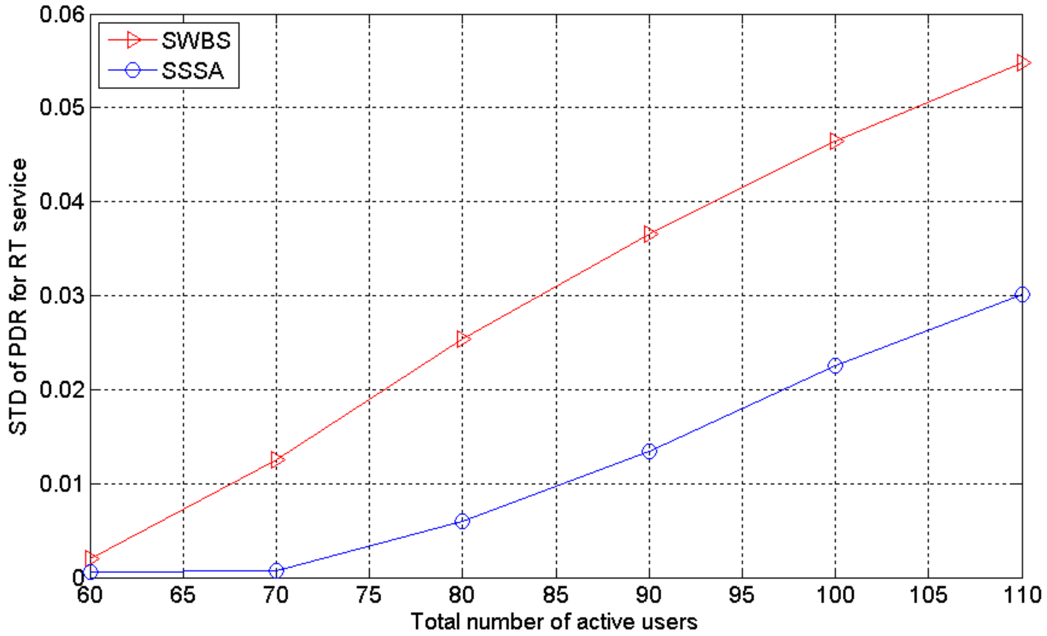


Figure 5-6 Standard deviation of average PDR distribution of RT users

The proposed SSSA shows lower variation in average PDR of RT users at all system loads as compared to SWBS algorithm. It is reduced by a significant percentage of 34.5% at a high system load of 110 active users. This is because the SSSA takes into account of HOL packet delay and queue length when prioritising users in the RT queue. By prioritising HOL packets, the average PDR due to time out, is reduced. The consideration of queue length significantly reduces average PDR variation among RT users by stabilising their queues. The STD between system loads of 60 to 70 active users remains constant because the system load is very low and delay requirements of all users are met, which results in near zero variation in average PDR values of RT users.

Figure 5-7 shows the average throughput of streaming service achieved by the proposed SSSA and SWBS under different system loads.

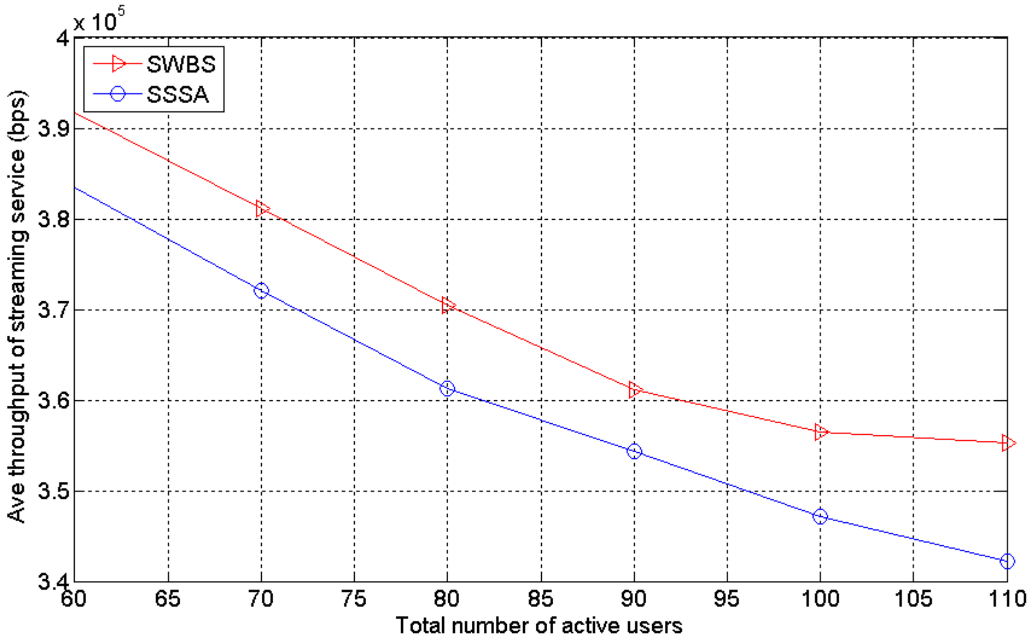


Figure 5-7 Average achieved throughput of streaming service

Both the proposed SSSA and SWBS algorithm meet throughput requirement of streaming service (240 kbps) at all system loads. However SSSA achieves slightly lower throughput than SWBS because it allocates sufficient resources to streaming and real-time services based on their QoS requirements and the rest to the background service, to prevent any starvation. SWBS does not differentiate between streaming and background services and treats them equally, which may results in either over allocation or starvation to background service.

The results on average achieved throughput by background service are also presented to verify that BE queue does not starve forever. This is the particular issue to be considered in communication systems, especially in LTE-A, which has diverse range of applications. Most of the current scheduling algorithms do not consider separately the background service while focusing on services with QoS requirements. That is why these algorithms either over allocate the background service or cause starvation to it. Although background service does not have QoS requirement but still it should not starve forever.

Figure 5-8 shows average achieved throughput of background service shown by SSSA and SWBS under different system loads.

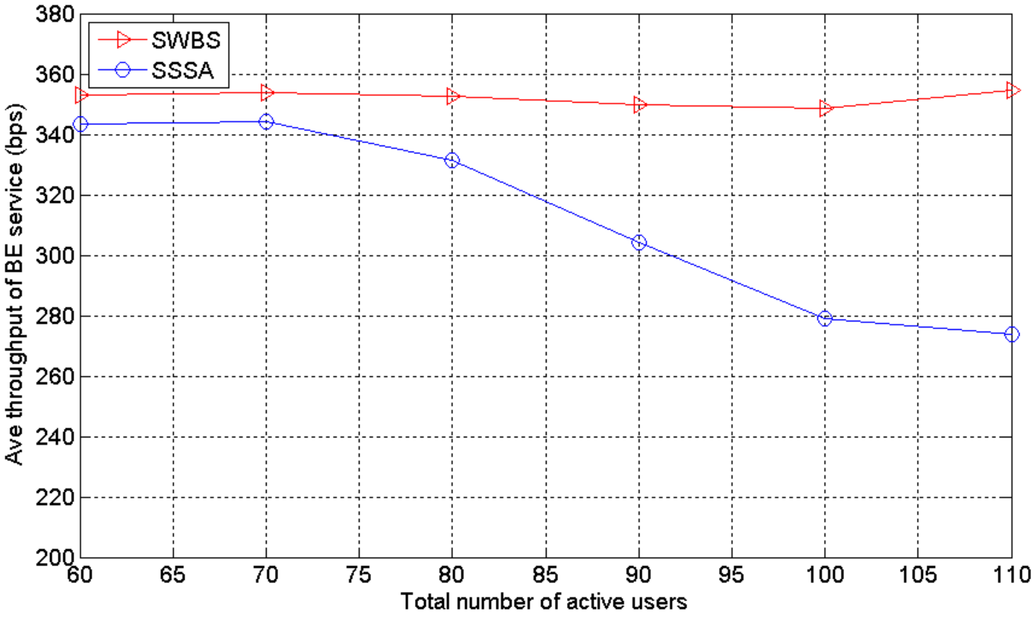


Figure 5-8 Average throughput of background service

The proposed SSSA shows lower throughput as compared to SWBS and the difference becomes higher when the system is highly loaded. This is because SSSA considers streaming and background as two separate services and tries to fulfil the minimum required throughput to streaming service first. SWBS however equally considers both services and achieves the average throughput of both at equal level. In this way it over allocates the BE queue and as a result cannot keep delay, delay variation and average PDR of real-time service lower, especially when system load is high. The proposed SSSA allocates resources to BE queue after considering QoS of RT and NRT queues.

5.3.2 System Performance

System performance is analysed by calculating *fairness among users* and overall *system throughput*. Fairness among users is calculated by Equation 3-24.

Figure 5-9 shows fairness achieved by the proposed SSSA, QoS aware SWBS algorithm and classic PF algorithm under different system loads.

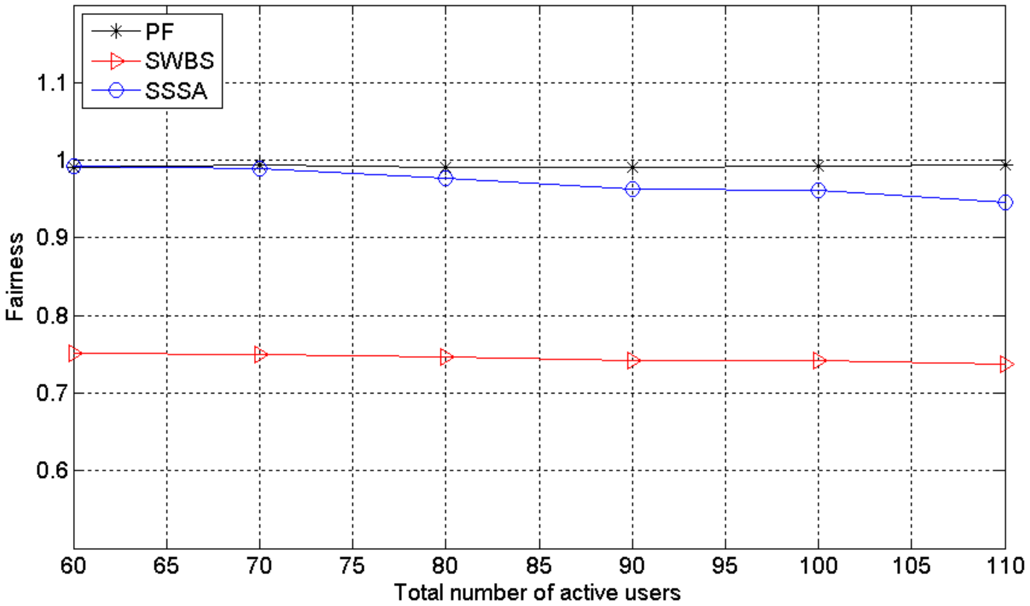


Figure 5-9 Fairness among all users

The proposed SSSA achieves higher fairness at all system loads, as compared to SWBS. This is because SSSA adaptively allocates resources to each queue and tries to prevent over allocation or starvation to any service. SWBS over allocates some services and cannot achieve good fairness. This is achieved by updating the priorities of users in each queue before scheduling decisions. The fairness results are also compared with classic PF algorithm, which is considered as a bench mark for a good trade-off between system throughput and user fairness. The proposed SSSA shows almost same fairness with PF at lower system loads from 60 to 70 active users because every user can get radio resource. At system load higher than 70 active users, fairness shown by SSSA becomes slightly lower than PF algorithm. This is because SSSA considers both user-level and system-level performance and PF algorithm considers only system-level performance.

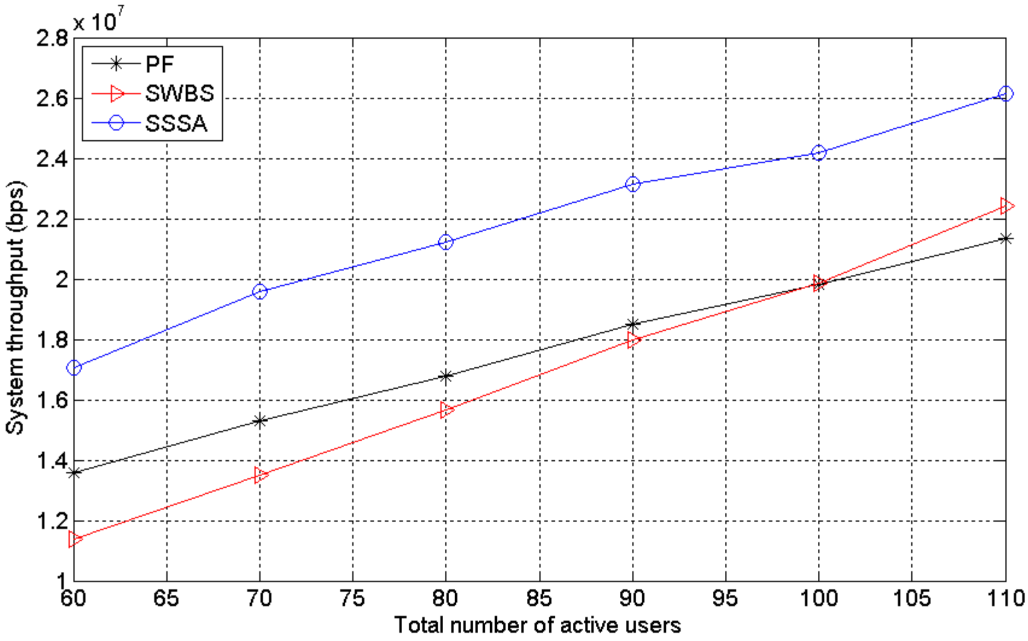


Figure 5-10 System spectral efficiency

Figure 5-10 shows system throughput achieved by the proposed QoS aware SSSA, SWBS and conventional PF algorithms under different system loads. The proposed SSSA outperforms all other algorithms. It improves overall system throughput by 4.35 Mbps at a system load of 100 active users, as compared to SWBS and MIX. This is because SSSA takes into account of both TD and FD MU diversity by considering radio channel conditions. It also shows good system throughput than classic PF algorithm, which considers both fairness and system throughput equally to make the best trade-off between them. QoS aware SWBS however shows lower throughput than PF algorithm up to a load of 90 active users and increases afterwards due to exploiting MU diversity in TD. As PF algorithm does not only improve system throughput but makes a balance between system throughput and user fairness and shows lower throughput than SWBS at higher system loads.

These results show that the proposed SSSA makes a better trade-off between user-level and system-level performance, as compared to SWBS and PF algorithms. SSSA meets the minimum throughput requirement of NRT traffic, and also reduces the average PDR variation in average PDR of RT users, average delay and average PDR of real-time service by sacrificing throughput of background service slightly. Meanwhile it maintains good system-level performance by improving system throughput and user fairness.

5.4 Joint (SSSA & ATDSA) Performance Evaluation

This thesis proposes a novel adaptive time domain scheduling algorithm (ATDSA), to allocate sufficient resources to real-time and streaming video service to meet their QoS requirements and allocate rest of the resource to background service, to maintain fairness among users of all services. Hebbian learning process is integrated in ATDSA to intelligently allocate available resources to different services. In addition K-mean clustering algorithm is used to further reduce average PDR of real-time service and the average PDR variation among RT users.

The joint results of the proposed SSSA and ATDSA are presented under different network scenarios. This is to validate the credibility of the proposed architecture under real network conditions. The focus of this research work is to improve the QoS of real-time voice and non-real time streaming video services while maintaining the overall system-level performance in terms system throughput and user fairness. Therefore, the simulation results include *average packet delay*, *variation in average PDR* of RT users and *average PDR* of real-time service, *minimum throughput* of non-real time service, *system throughput* and *user fairness*.

The performance of the proposed ATDSA is compared against two recent QoS aware MIX [JNTMM08] and SWBS algorithms [SYLX09].

The system model used in this section is shown in Figure 5-11.

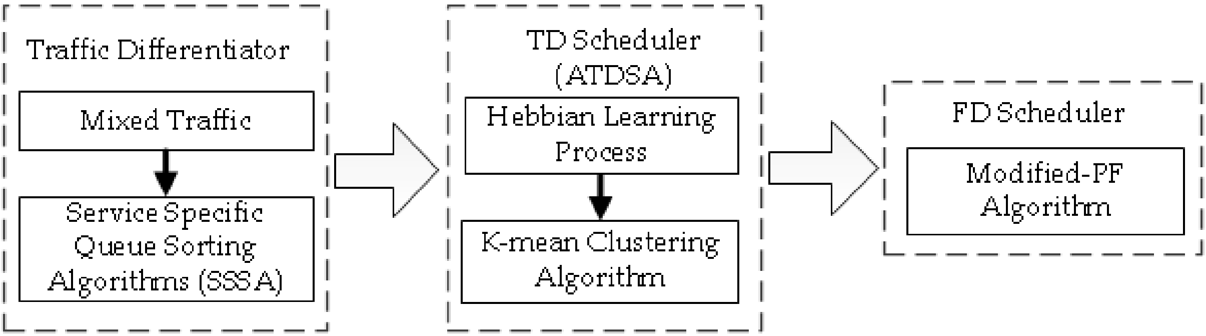


Figure 5-11 System model of CLPSA

It considers all proposed algorithms at different stages of CLPSA, which include SSSA at the *Traffic Differentiator*, ATDSA at the *TD Scheduler* and M-PF at the *FD Scheduler* stage.

To exemplify real network, the performance of the proposed scheduling architecture CLPSA is evaluated under the following simulation scenarios.

Scenario 1: Equal distribution of real-time and non-real time services.

(Real-time service 50%, streaming service 25%, Background service 25%)

Scenario 2: Real-time service dominant over other services.

(Real-time service 70%, streaming service 20%, Background service 10%)

Scenario 3: Background service dominant over other services.

(Real-time service 20%, streaming service 20%, Background service 60%)

Scenario 4: Streaming service dominant over other services.

(Real-time service 20%, streaming service 70%, Background service 10%)

5.4.1 Scenario 1: Equal Distribution of Real Time and Non Real time Services

In this scenario, real-time service is 50%, streaming service is 25% and background service is 25% of the total system load. The performance of the proposed scheduling architecture is represented by CLPSA, which is compared with SWBS and MIX.

Figure 5-12 shows average packet delay for the proposed CLPSA and the reference (MIX and SWBS) algorithms under different system loads.

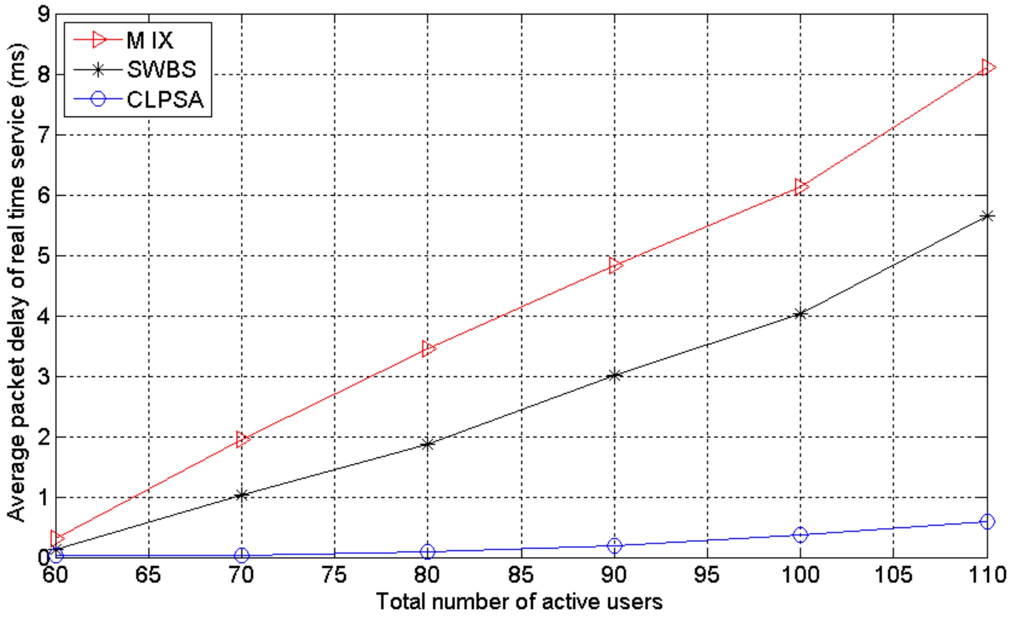


Figure 5-12 Average packet delay of real-time service

As expected, average packet delay of real-time service for all algorithms increases with the number of active users however the proposed CLPSA shows the lowest delay. This is because CLPSA uses a novel delay-dependent queue-sorting algorithm for real-time service, which prioritises users with higher HOL packet delay and longer queues. It uses novel ATDSA in the TD, which allocates resources to real-time service based on its QoS requirement. It allocates sufficient resources to further reduce average packet delay. MIX due to lack of service-specific queue-sorting and uses fair scheduling scheme in the TD, which is why it cannot keep average packet delay low at all system loads. The SWBS algorithm however due to its QoS aware scheduling scheme shows lower packet delay as compared to MIX.

Figure 5-13 shows the average PDR for real-time service for all algorithms under different system loads.

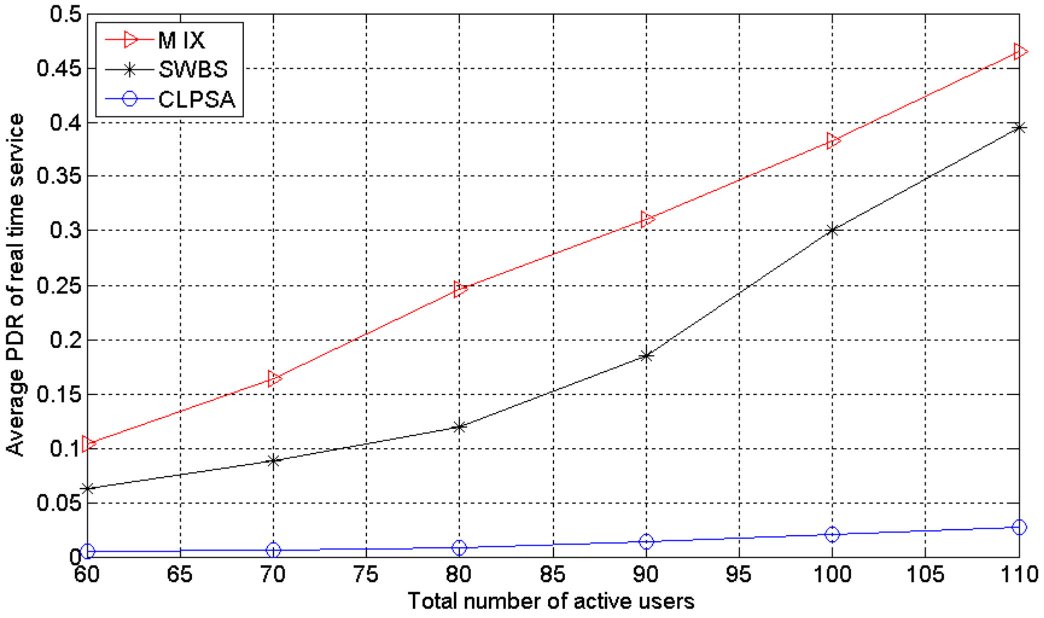


Figure 5-13 Average PDR of real-time service

The proposed CLPSA shows the lowest average PDR at all system loads. It is lower by a value of 0.32 and 0.42 than SWBS and MIX, respectively at a highly loaded scenario having 110 active users. This is due to the adaptive scheduling scheme used by CLPSA in the TD. It takes information on average PDR of real-time service to take scheduling decisions. CLPSA adaptively increases resource allocation to real-time service when average PDR is higher, thus, shows lower average PDR values with the system load. MIX shows the highest average PDR at all system loads and SWBS because of its service-specific scheduling scheme, shows lower average PDR than MIX.

Figure 5-14 shows the variation (STD) in average PDR of all RT users under different network loads.

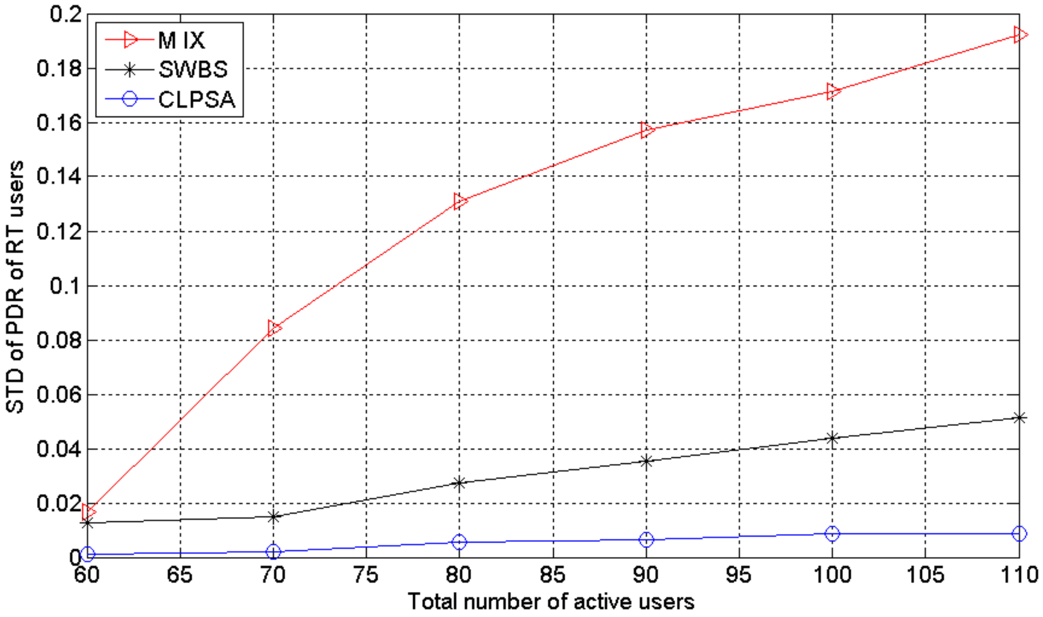


Figure 5-14 Standard deviation of average PDR of RT users

The proposed CLPSA achieves the lowest value of STD of average PDR of all RT users with system load, as compared to MIX and SWBS. This is because of the incorporation of K-mean clustering in CLPSA, which re-arranges the priority of RT users based on their average PDR values. In this way average PDR of individual RT users is significantly reduced resulting in lower variation (STD) in average PDR of RT users. The integration of K-mean clustering further reduces STD as compared to SSSA alone, which is shown in Figure 5-6. Therefore PDR fairness among RT users is increased significantly.

QoS aware SWBS has lower STD values than MIX because SWBS considers packet delay while taking scheduling decisions. Meanwhile MIX has no delay-dependent strategies for real-time service thus shows the highest variation in PDR.

Figure 5-15 shows the average throughput of streaming service achieved by MIX, SWBS and CLPSA algorithms with system load.

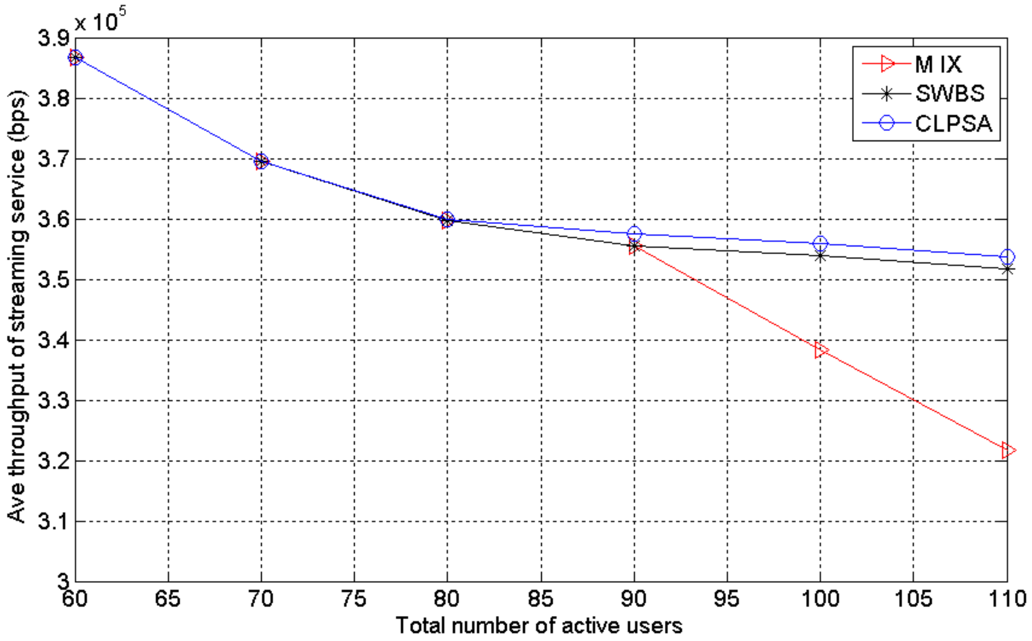


Figure 5-15 Average achieved throughput of streaming service

All algorithms achieve almost equal average achieved throughput up tp a system load of 80 active users. However when the load is further increased, the proposed CLPSA outperforms SWBS and MIX algorithms, by achieving the highest throughout. This is because the proposed CLPSA uses service specific queue-sorting algorithm SSSA for prioritising users demanding streaming video service. The SSSA prioritises users with lower average achieved throughput to improve QoS of NRT users. The adaptive scheduling algorithm in the TD, ATDSA allocates sufficient resources to meet throughput requirement of streaming service. However MIX due to its fair scheduling scheme in the TD cannot allocate resources to all NRT users at higher system load therefore its average achieved throughput drops when system load is higher than 90.

Figure 5-16 shows the fairness achieved by MIX, SWBS and the proposed CLPSA under different system loads.

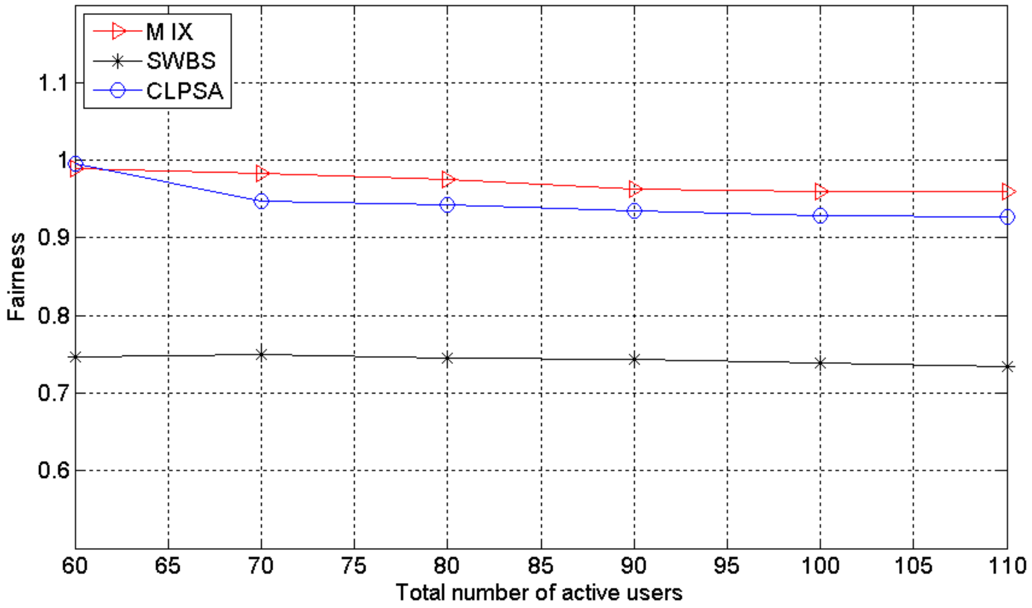


Figure 5-16 Fairness among all users

MIX due to its fair scheduling policy shows the highest fairness in this scenario where real-time and non-real time services have equal load. The proposed CLPSA shows slightly lower fairness than MIX, which is because CLPSA considers QoS of all services along with fairness. SWBS however shows the lowest fairness because it only focuses on the QoS provision to real-time and non-real time services. The good fairness index shown by the proposed CLPSA is due to its particular design of allocating just enough resources to QoS demanding services and the rest to background service. This is achieved by integrating Hebbian learning process in CLPSA, which adaptively allocates sufficient radio resources to real-time and streaming services and the rest to background service.

Figure 5-17 shows the overall system throughput achieved by all algorithms under different network loads.

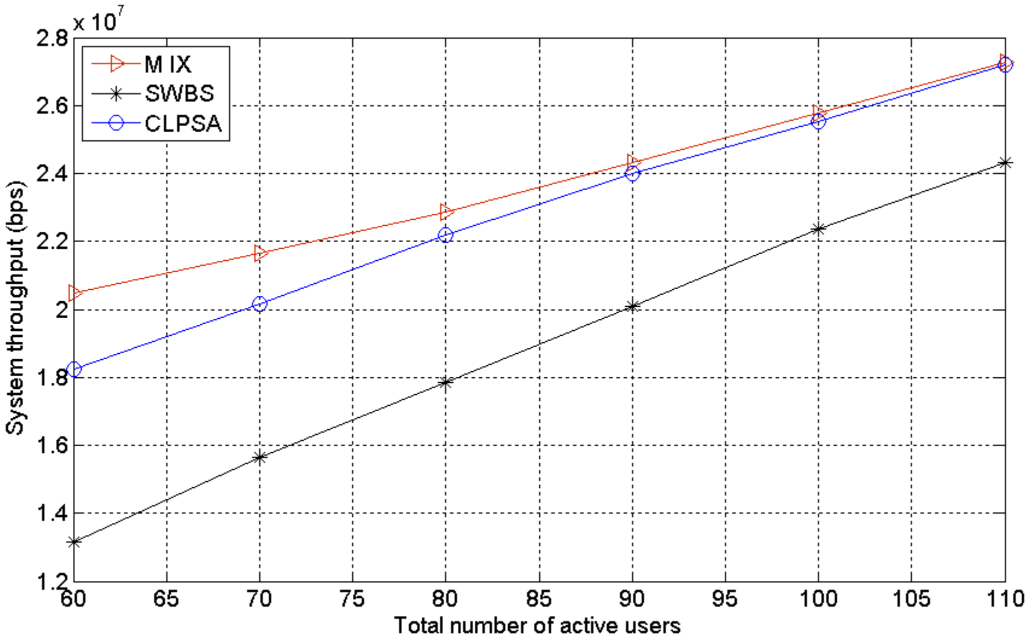


Figure 5-17 System throughput

MIX shows the highest throughput due to its channel-dependent queue-sorting algorithms. The proposed CLPSA achieves 2% lower throughput than MIX at low system load of 60 active users, which gradually increases at system loads higher than 60 active users and becomes almost equal to MIX at system load of 110 active users. This is because the proposed CLPSA takes into account of delay and throughput requirements of real-time and streaming service, respectively in queue-sorting algorithms rather than only using channel dependent queue-sorting algorithms, as MIX does. In addition it considers fairness along with system throughput and tries to allocate a fair share of radio resources to all services by using ATDSA. At lower system loads, as the number of users are small, every user gets equal resources increasing fairness. As increase in fairness causes a decrease in system throughput so the proposed CLPSA shows slightly lower throughput performance at lower system loads as compared to MIX. SWBS basically focuses QoS provision thus it achieves lower system throughput as compared to both MIX and CLPSA.

5.4.2 Scenario 2: Real Time Service Dominant over Other Services

In this scenario, real-time service is 70%, streaming video service is 20% and background service is 10% of total system load.

Figure 5-18 shows average packet delay of real-time service under different system loads.

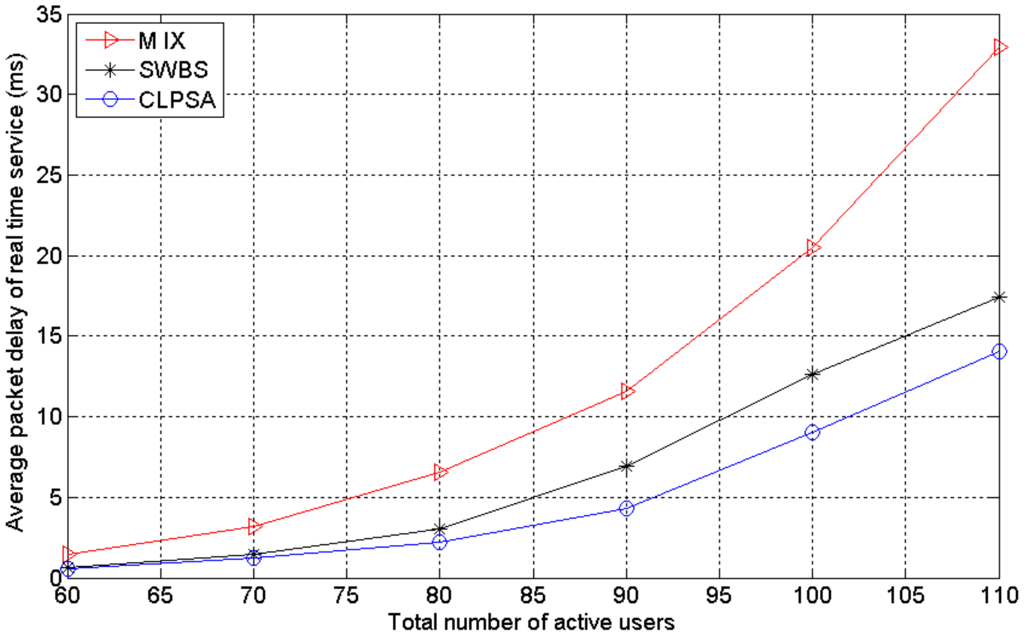


Figure 5-18 Average packet delay of real-time service

As expected, average packet delay increases for all algorithms with the system load however the proposed CLPSA shows the lowest delay. This is because CLPSA uses delay-dependent queue-sorting algorithm for real-time service, which prioritises users with higher HOL delay and longer queues. Also CLPSA uses adaptive TD scheduling algorithms, which adaptively allocates resources to different traffic types to meet their QoS requirements under changing network conditions. SWBS shows almost similar packet delay to CLPSA at lower system loads of 60 and 70 active users however the difference becomes higher at higher system loads above 70 active users. MIX shows the highest packet delay out of all algorithms.

The average PDR of real-time service is shown in Figure 5-19 under different system loads.

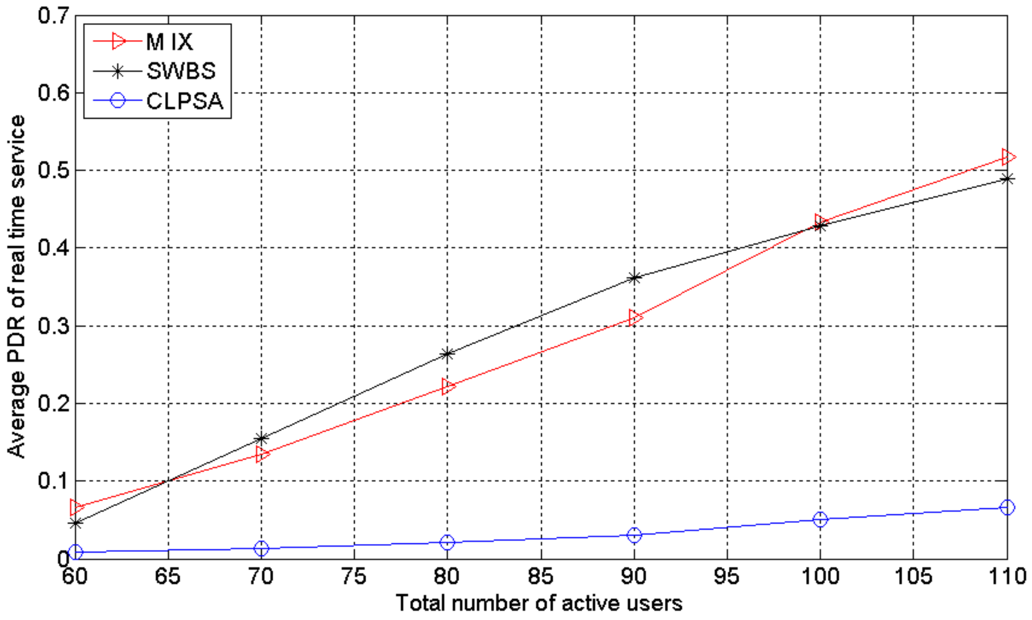


Figure 5-19 Average PDR of real-time service

The proposed CLPSA shows the lowest average PDR values at all system loads because of its adaptive scheduling policy, which takes information on average PDR of real-time service before each scheduling decision. It increases resource allocation to real-time service when average PDR becomes higher. MIX and SWBS have significantly higher average PDR than CLPSA and it increases in a near linear way with system load. The reason is that MIX and SWBS do not take into account of the information on average PDR of real-time service before taking scheduling decisions and are not adaptive to the changing network scenarios.

Figure 5-20 shows STD of average PDR of all RT users under increasing system load.

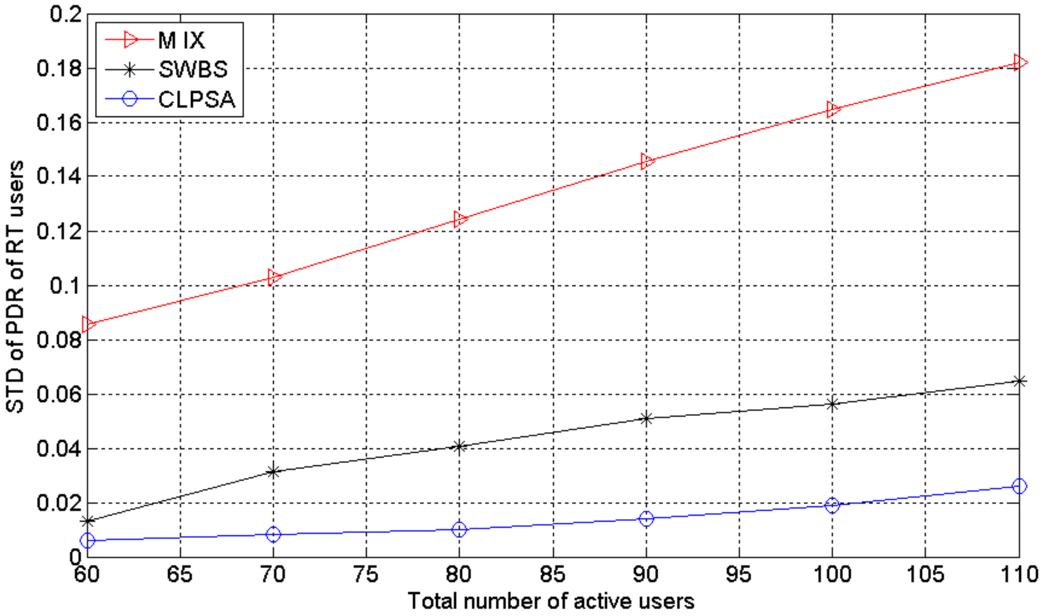


Figure 5-20 Standard deviation of average PDR distribution of all RT users

The proposed CLPSA outperforms all reference algorithms by showing the lowest STD at all system loads. This is because of K-mean clustering algorithm, which prioritises users with higher average PDR values, thus, reducing the STD of average PDR distribution of all RT users. However in this scenario where real-time service is dominant, STD shown by CLPSA is slightly higher than STD achieved in the previous scenario where real-time and non-real time services have equal load, as shown in Figure 5.13. This is because under this scenario real-time service is the dominant. Mix shows the highest STD when compared with CLPSA and SWBS. This is because MIX does not use service-specific queue-sorting and uses fair scheduling to allocate radio resources to users demanding different services.

The average achieved throughput for streaming service is shown in Figure 5-21 for the proposed CLPSA, SWBS and MIX.

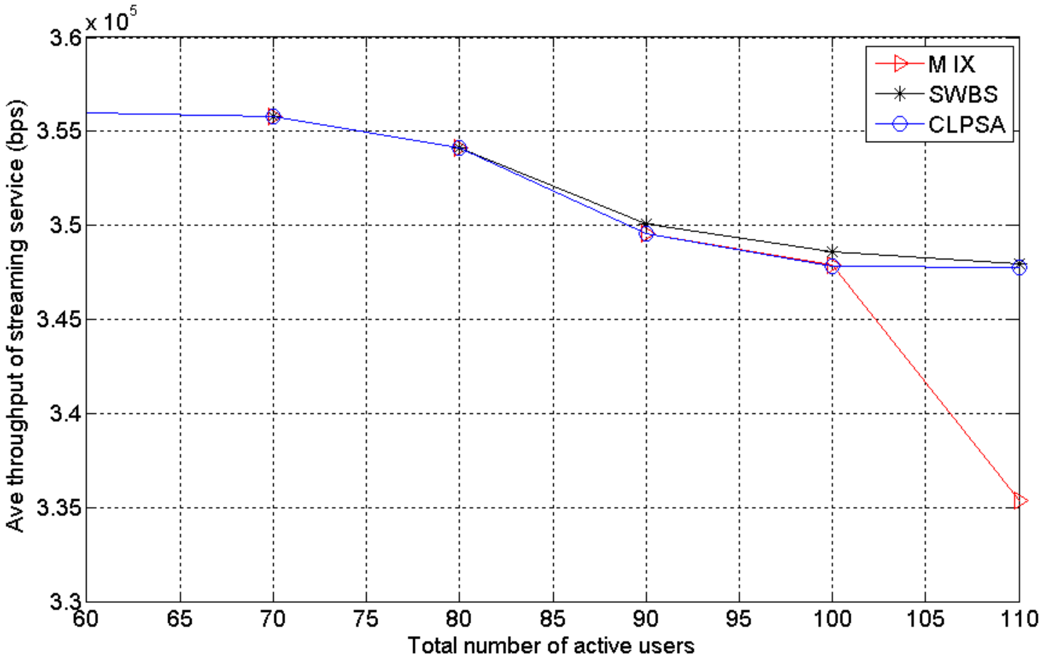


Figure 5-21 Average achieved throughput of streaming service

All algorithms meet the throughput requirements of streaming video service because users demanding this service are only 20% of whole active users in this scenario, where real-time service is dominant over all other services. MIX however shows a sudden drop in average achieved throughput at higher system loads because of its fair scheduling scheme. However due to small number of users demanding streaming service, it can still meet minimum throughput requirements at all system loads.

Figure 5-22 shows fairness achieved by all algorithms under different system loads.

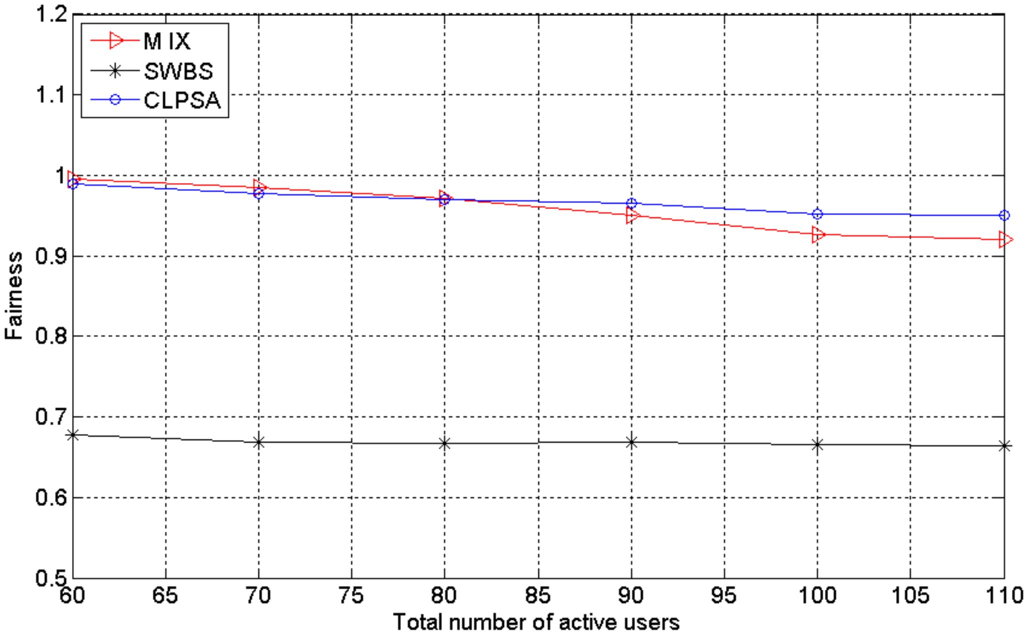


Figure 5-22 Fairness among all users

The proposed CLPSA maintains fairness at good level with system load. This is because of its design to prevent over allocation to QoS demanding services and service starvation of background service, by using Hebbian learning process in CLPSA. It adaptively allocates sufficient resources to QoS demanding services and the rest to the background service, thus, improves fairness. MIX due to its fair scheduling scheme achieves slightly higher fairness than CLPSA at lower system loads however it drops at higher system loads. SWBS shows significantly lower fairness as compared to MIX and CLPSA. This is because SWBS mainly considers QoS provision to different services.

Figure 5-23 shows the system throughput achieved by all algorithms under variable system loads.

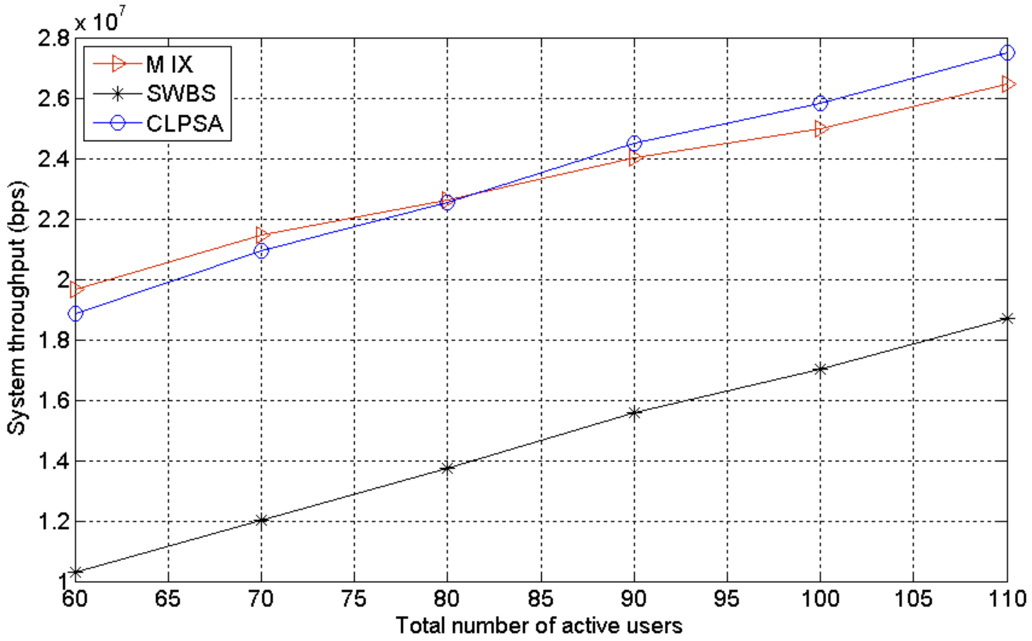


Figure 5-23 System throughput

The proposed CLPSA achieves almost similar system throughput at all system loads as achieved by MIX, which uses channel dependent queue-sorting algorithms. At system loads lower than 80 active users, CLPSA shows slightly lower system throughput, which is because of considering both user-level and system-level performance at the same time. However at system loads higher than 80 active users, CLPSA achieves higher system throughput than MIX.

SWBS shows a nearly linear increase in system throughput with the system load, which is lower than both MIX and CLPSA algorithms by a significant value of 60.18 Mbps at almost all system loads.

5.4.3 Scenario 3: Background Service Dominant over Other Services

In this scenario, the background service is 60%, real-time service is 20% and non-real time service is 20% of total system load.

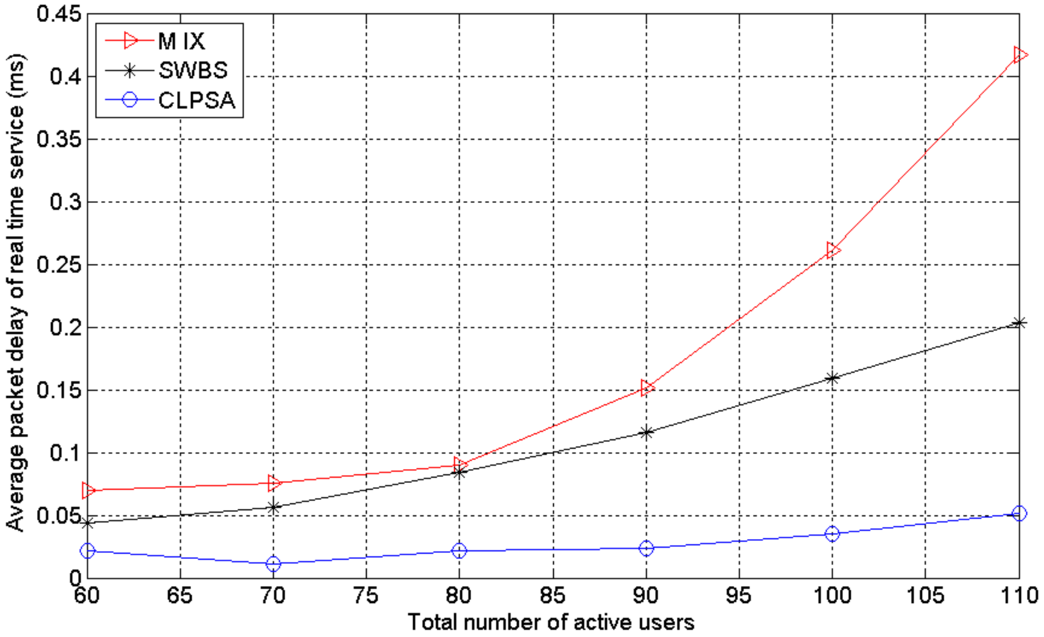


Figure 5-24 Average packet delay of real-time service

Figure 5-24 shows the comparison of the average packet delay of the real-time service between the proposed CLPSA, MIX and SWBS algorithms with system load. The average packet delay of the real-time service increases with the system load for these three algorithms however, the proposed CLPSA shows the least packet delay. This is because CLPSA uses delay-dependent queue-sorting algorithms for real-time service which prioritises users with higher HOL packet delay and longer queues. Adaptive scheduling with Hebbian learning process in ATDSA, allocates sufficient resources to real-time service to meet its QoS requirement under this scenario as well. MIX shows the highest packet delay with an average value of 0.44 ms at a system load of 110 active users. This is because MIX lacks delay-dependent queue-sorting algorithms as used by CLPSA. SWBS shows lower delay than MIX as it uses QoS aware scheduling schemes to guarantee QoS to real-time and non-real time services.

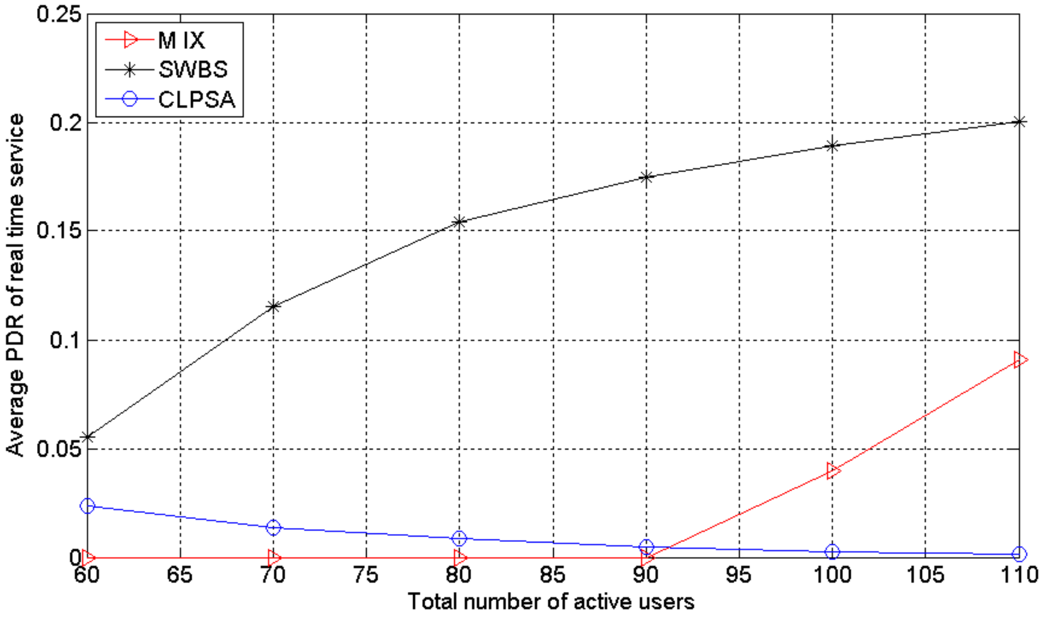


Figure 5-25 Average PDR of real-time service

The average PDR of real-time service with system load is shown in Figure 5-25. The proposed CLPSA shows average PDR which is lower than the PDR threshold. MIX shows zero average PDR up to a load of 90 active users and increases abruptly afterwards. This is due to fair scheduling scheme in MIX. In fair scheduling, one user is selected from each queue to be scheduled. At lower system loads, with this strategy, RT queue gets enough resource to satisfy QoS of all real-time users resulting in zero average PDR. SWBS shows the highest average PDR. CLPSA shows slightly higher average PDR at lower system loads because it provides just enough resources to keep average PDR under threshold. However at higher system loads, it increases resource allocation to real-time service adaptively based on Hebbian learning process resulting and keeps average PDR at the low level.

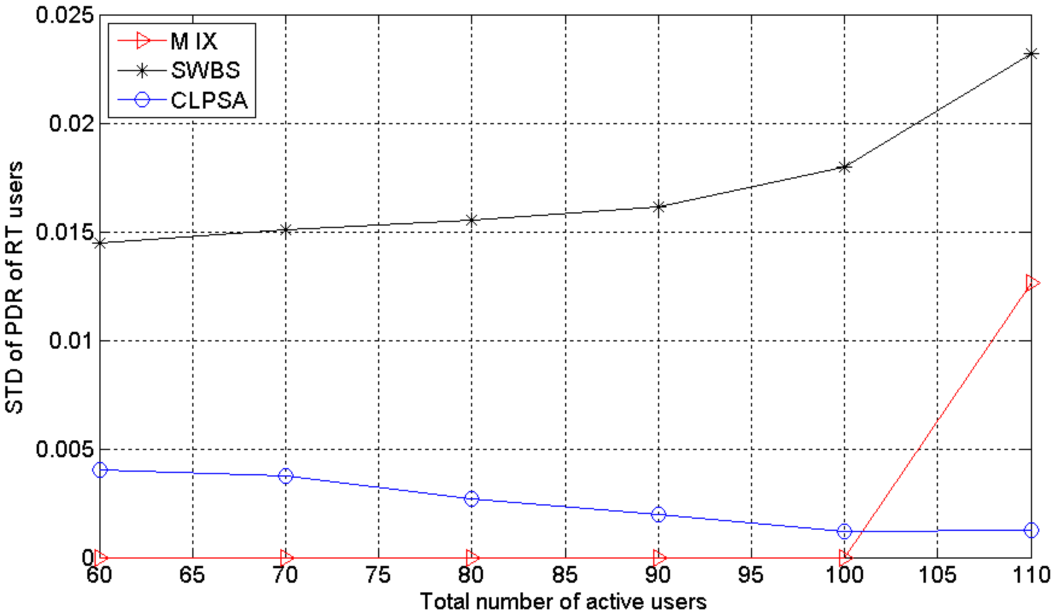


Figure 5-26 Standard deviation of average PDR distribution of all RT users

Figure 5-26 shows STD (variation) of average PDR of all RT users under different network loads. For this scenario where real-time service is only 20% of the total load, all algorithms show lower STD. However MIX shows zero STD value for system loads up to 100 active users but it increases abruptly when load is further increased above 100 active users. This is because MIX schedules users one-by-one from each queue and allocates a fair share of resources to each traffic type. As in this particular scenario, the real-time users are only 20% of all active users, thus with one-by-one scheduling policy up to a load of 100 active users, all real-time users get enough resources in each TTI and hardly suffer any PDR. However at higher system load, MIX is unable to schedule all real-time users resulting in a sudden increase in average PDR, which is not the case with the proposed CLPSA. Although CLPSA shows slightly higher STD than MIX at lower system loads but it maintains lower STD at higher system loads as well. This is due to K-mean clustering algorithm, which prioritises users with higher PDR values. In addition, the proposed CLPSA by using its adaptive scheduling scheme allocates sufficient resources to real-time service at all system loads.

However at lower system loads, CLPSA prevents extra allocation to real-time service, as MIX does.

In this scenario, CLPSA shows the lowest STD as compared to the previous scenarios, which are shown in Figure 5-14 and Figure 5-20 because real-time service is only 20% of total load. SWBS shows the highest STD because it allocates most of the resources to streaming service when system is loaded with streaming service.

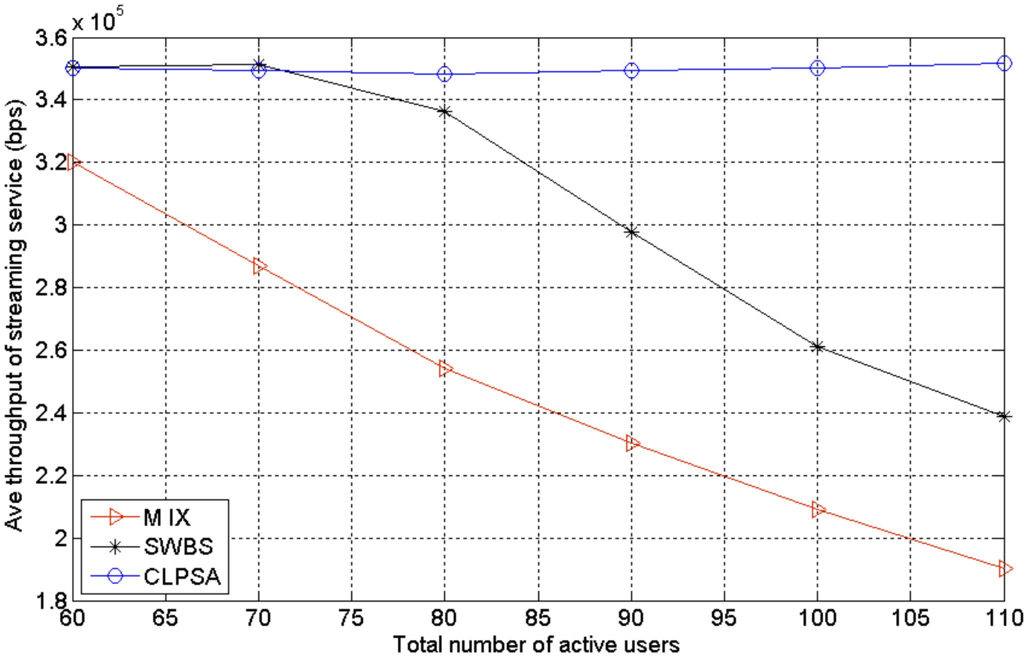


Figure 5-27 The average achieved throughput of streaming traffic

The average achieved throughput of streaming service is shown in Figure 5-27 under different system loads. The proposed CLPSA shows the highest average throughput values at all system loads. This is because in this particular scenario, CLPSA due to its intelligent techniques meets the QoS requiremnets of real-time voice and non-real time streaming video service first,only then assigns the remaining resource to background service which is dominant. SWBS does not differenciate between background and streaming video services and takes both as non-real time traffic with certain throughput requirements. Therefore allocates most of resources to background service when it is dominant. And MIX due to lack of service-specific queue-sorting can not meet QoS of streaming video service.

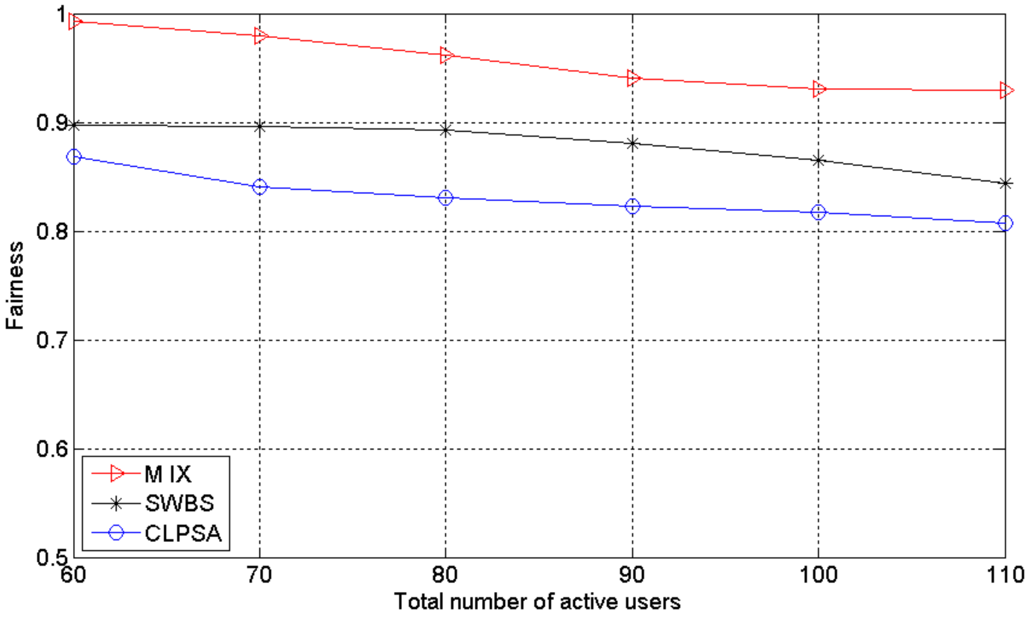


Figure 5-28 Fairness among all users

The fairness among users is calculated using Equation 5-1 for all these algorithms and the results are presented in Figure 5-28. MIX uses fair scheduling algorithm and achieves the highest fairness and SWBS uses equal prioritises for the real-time and streaming service and achieves fairness slightly higher than CLPSA. As CLPSA adaptively allocates resources to the real-time and streaming services based on achieved QoS measures, it loses slightly on the fairness as compared to MIX and SWBS only in this particular scenario where background service is dominant. This is because it prioritises QoS demanding services over background service and first tries to meet their QoS demands. However it still maintains the fairness at reasonably good value of 0.87 at system load of 60 active users and 0.81 at system load of 110 active users.

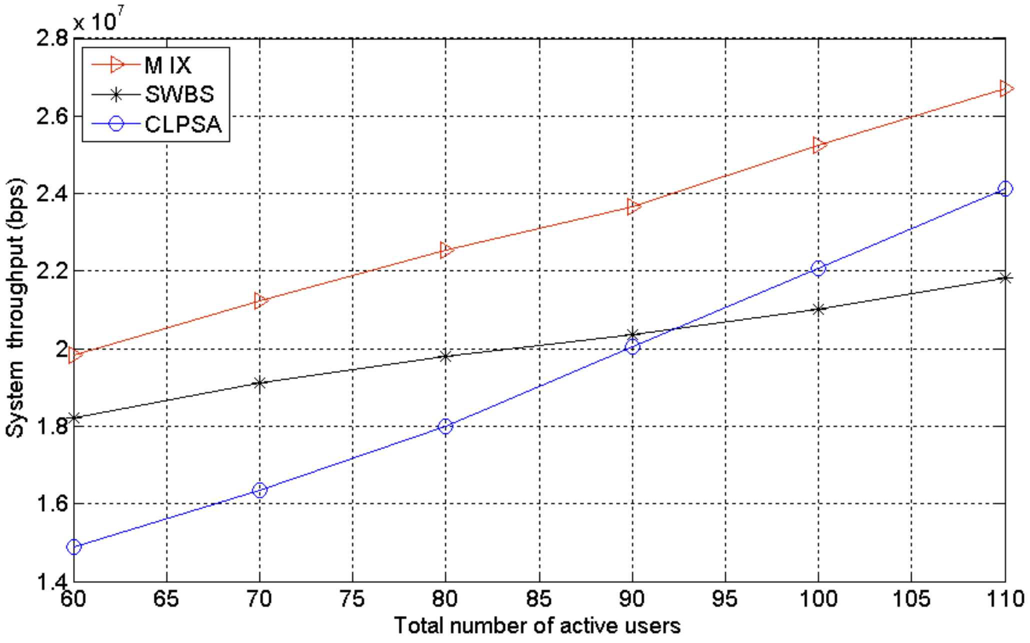


Figure 5-29 System throughput

System throughput is the sum of throughput of all users and is shown in Figure 5-29 under different system loads. MIX shows the highest throughput because it only uses channel aware MAX C/I and PF as queue-sorting algorithms. SWBS shows good performance as compared with the proposed CLPSA up to a system load of 95 active users and then becomes lower than CLPSA. This is because SWBS does not consider streaming and background service separately. The proposed CLPSA gives priority to voice and streaming services as compared with background service. It only allocates resources to background service when the throughput requirements of streaming service are met. As in this scenario the background service is dominant, CLPSA shows slightly lower throughput than SWBS at lower system loads from 60 to 95 active users however it becomes higher than SWBS at higher system loads from 95 to 110.

5.4.4 Scenario 4: Streaming Service Dominant over Other Services

In this scenario, streaming service is 70%, real-time service is 20% and background service is 10% of total system load.

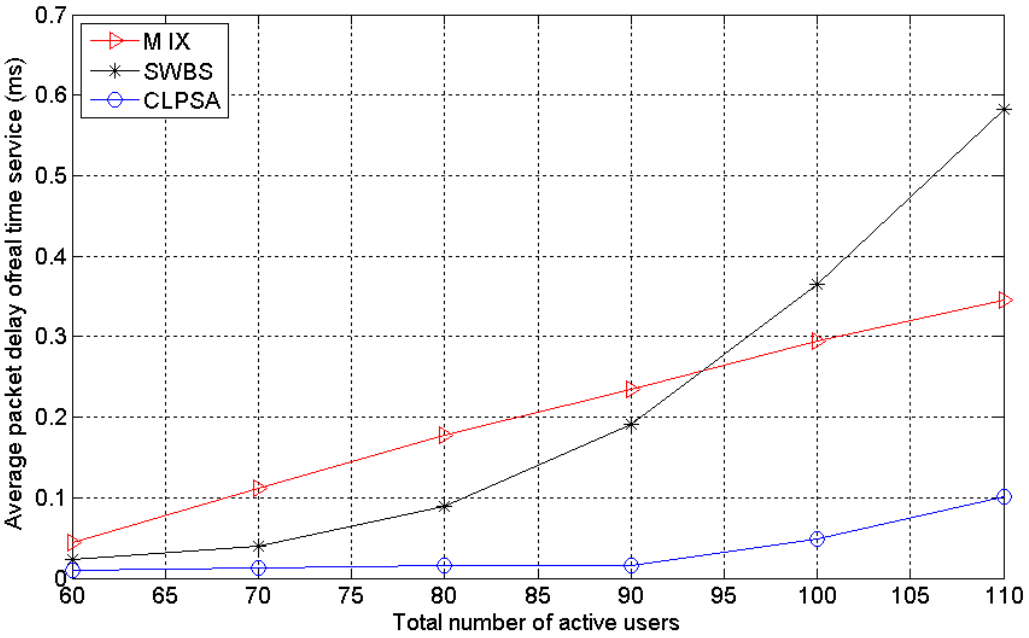


Figure 5-30 Average packet delay of real-time service

The average packet delay for real-time service is shown in Figure 5-30 for MIX, SWBS and CLPSA algorithms under different network loads. The proposed CLPSA shows the lowest packet delay due to its delay-dependent queue-sorting algorithm. It prioritises users with higher HOL packet delay and longer queues and reduces average packet delay. The integration of Hebbian learning further reduces average packet delay of real-time service by allocation sufficient resources to meet QoS requirement under changing network conditions. MIX shows a near linear increase in average delay with system load. This is because MIX uses fair scheduling scheme and allocates resources to all users equally. QoS aware SWBS shows significant increase in packet delay with the system load. At lower system load, SWBS can meet QoS of both RT and NRT queue but at higher system load in this scenario where NRT queue is dominant, it allocates more resource to NRT queue resulting in higher packet delay to RT queue. It therefore shows higher packet delay even than MIX algorithm at higher system load.

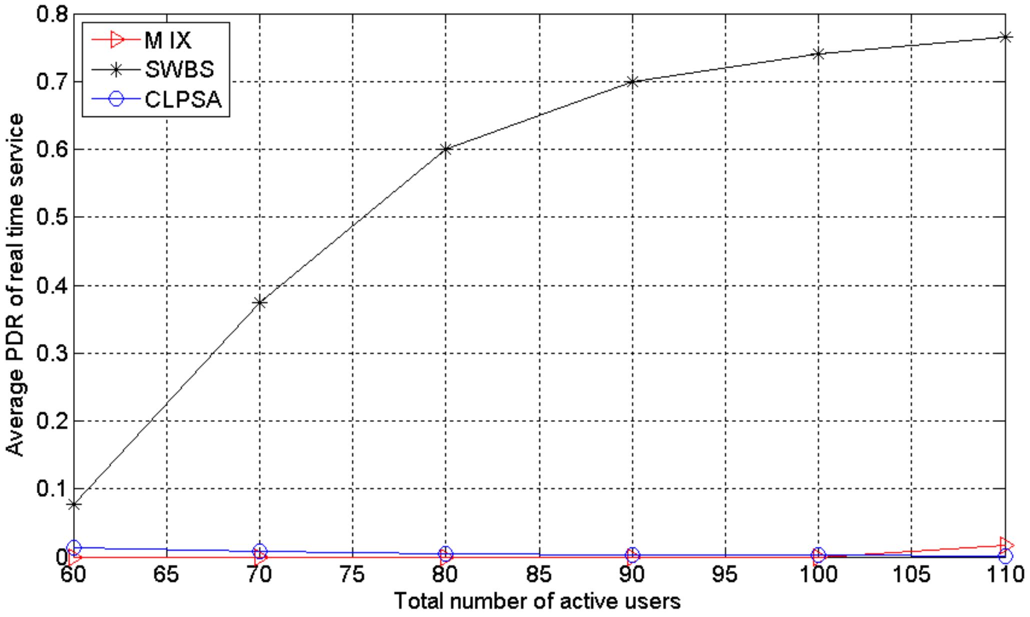


Figure 5-31 Average PDR of real-time service

Average PDR of real-time service is shown in Figure 5-31 for all algorithms under different system loads. The proposed CLPSA outperforms the reference SWBS algorithm at all system loads and keeps average PDR under the PDR threshold, which is 0.1 according to 3GPP [3G))12a]. This is because CLPSA uses Hebbian learning process which adaptively increases resource allocation to real-time service when average PDR of real-time service increases. SWBS does not take scheduling decisions adaptive to the information on average PDR information. MIX algorithm however shows very low average PDR in this scenario when real-time service is only 20% of the total system load. This is because MIX uses fair scheduling scheme in the time domain. With fair scheduling each user in RT queue get enough radio resource in each TTI resulting in the lowest average PDR.

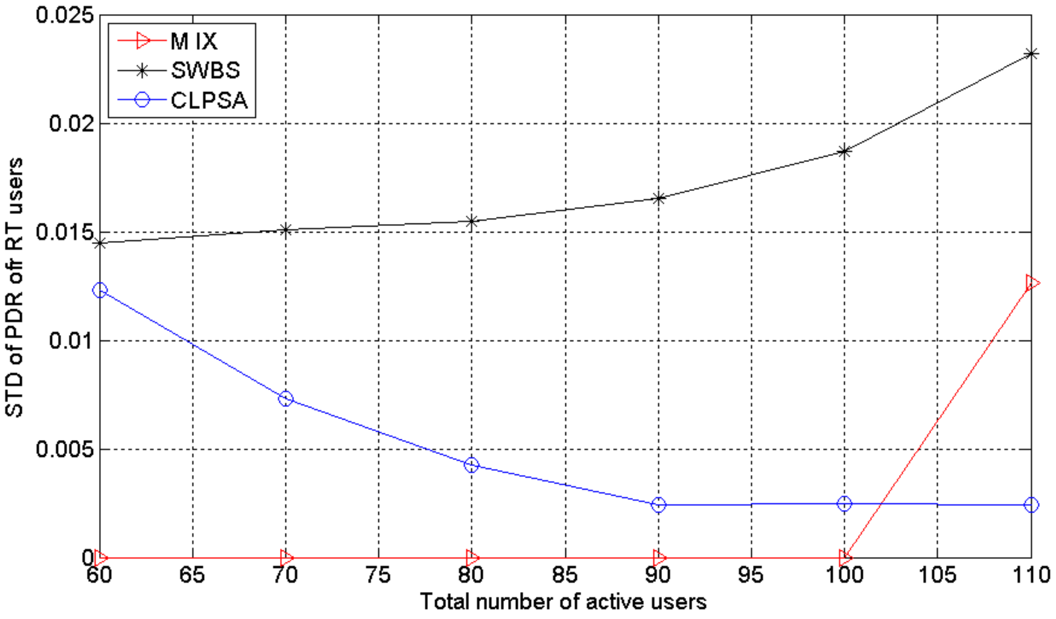


Figure 5-32 Standard deviation of average PDR distribution of RT users

Figure 5-32 shows STD of average PDR of all RT users with different network loads. As the number of the real-time users in this scenario is only 20% of the total system load, all algorithms have average PDR lower than the PDR threshold. Similar to scenario 3 where RT users are 20% of the total system load, MIX shows the least STD up to a load of 100 active users by its fair scheduling but cannot maintain it at higher system loads. The proposed CLPSA however maintains STD at lower level at higher system load as well by intelligently prioritising RT users.

MIX uses fair scheduling scheme and it has zero STD up to a load of 100 active users and it increases suddenly afterwards. This is because it allocates resources to all users equally without considering the changing load by different services. The proposed CLPSA considers the changing network scenarios and takes scheduling decisions accordingly. It shows slightly higher STD at lower system loads up to 104 active users however, it maintains STD at lower level at higher system loads from 95 to 110 active users. This is because CLPSA prioritises RT users based on the information of average PDR at the individual user-level, to keep average PDR of all RT users lower than PDR threshold. This is achieved by K-mean clustering algorithm which prioritises users with higher average PDR. SWBS however shows the highest STD at all system loads comparing with MIX and CLPSA.

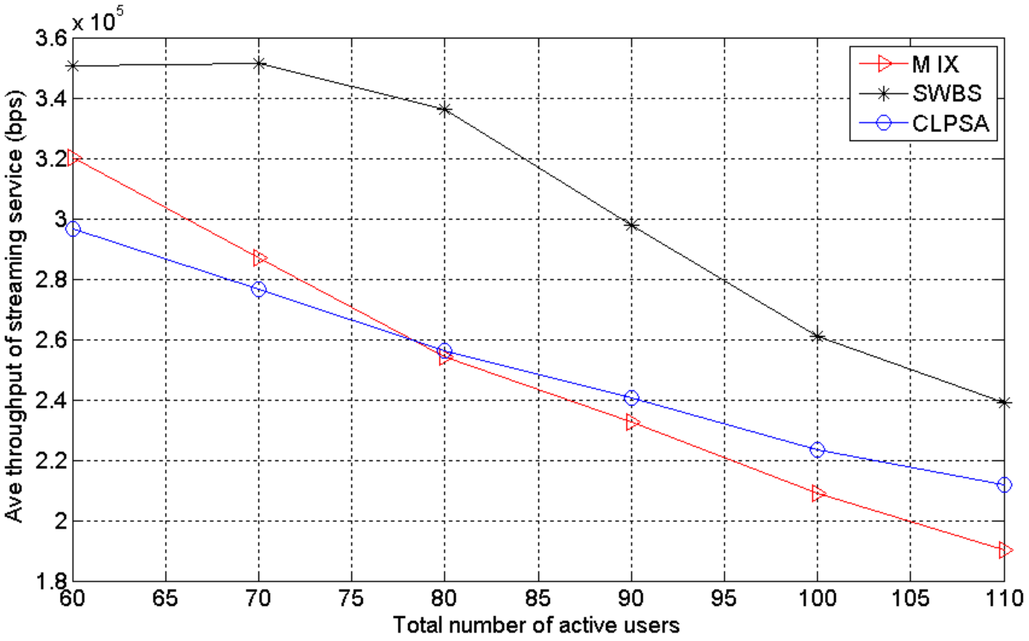


Figure 5-33 Average achieved throughput of streaming traffic

Figure 5-33 shows average achieved throughput for streaming video service under different system loads. SWBS shows the highest average achieved throughput as it allocates most of the radio resource to streaming video at the cost of lower QoS provision to real-time service, which is not the case in CLPSA. The proposed CLPSA tries to keep a balance among all services by adaptive resource allocation policy, which is implemented in CLPSA by integrating Hebbian learning process (Section 3.4.1). It adaptively allocates sufficient resources to meet QoS of real-time and non-real time services and assigns the rest to background service. It thus, shows slightly lower throughput in this scenario where streaming service is dominant. MIX because of its fair scheduling, shows good average achieved throughput than CLPSA at lower system loads but cannot maintain it at higher system loads. At lower system load, CLPSA shows lower performance than MIX because

CLPSA is designed to allocate just enough resource to meet QoS requirements. MIX however is blind of QoS requirement and the changing network scenarios.

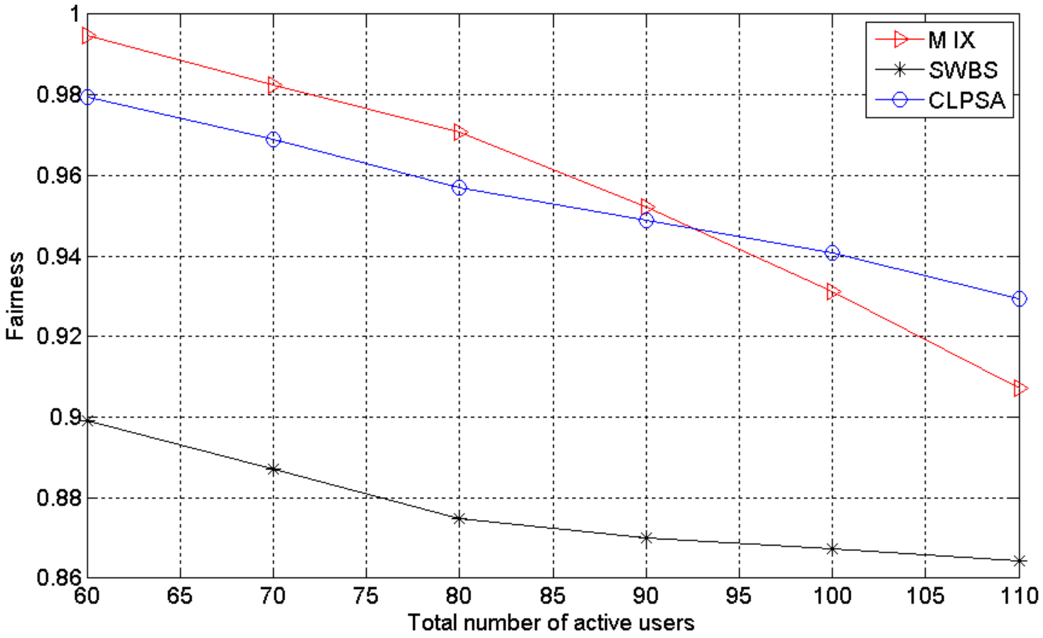


Figure 5-34 Fairness among all users

The fairness index achieved by MIX, SWBS and CLPSA algorithms is shown in Figure 5-34 under different system loads. MIX achieves the highest fairness because of its fair scheduling scheme however it cannot maintain it at higher system load due to massive streaming video service. Despite its fair scheduling scheme, Mix due to uneven distribution of services in this scenario cannot maintain the highest fairness under higher system loads. CLPSA is capable to maintain good fairness level due to its adaptive policy in the TD. It achieves fairness which is slightly lower than MIX at lower system loads but its fairness exceeds that of MIX at higher system loads due to its adaptive policy. SWBS however shows the least fairness because it focuses more on QoS provision and lacks fairness criteria in its design.

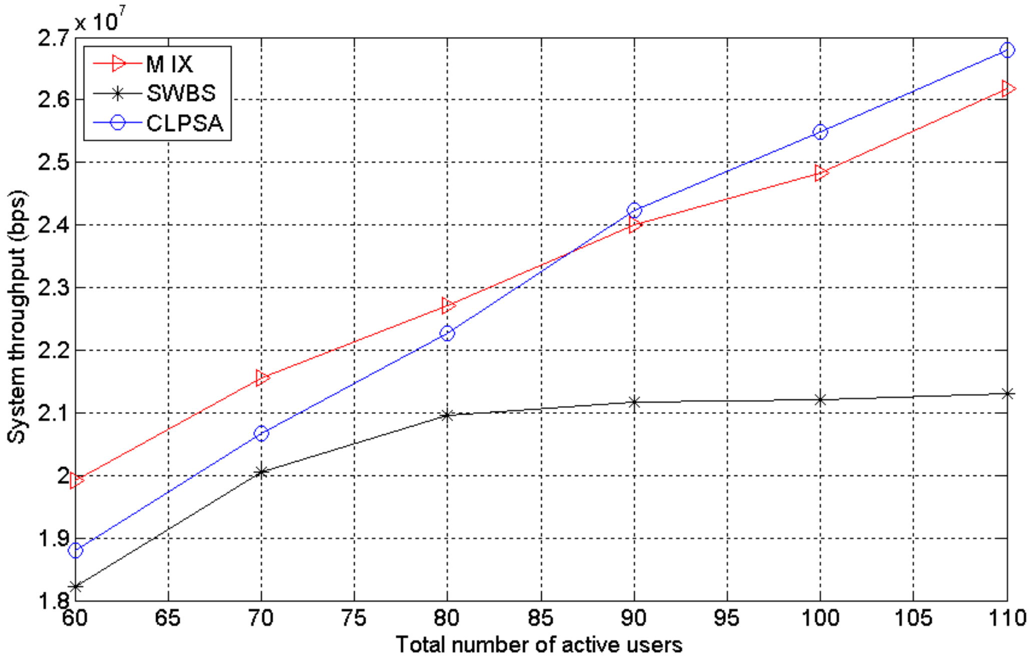


Figure 5-35 System spectral efficiency

Figure 5-35 shows the system throughput for all algorithms under different system loads. The proposed CLPSA shows slightly lower system throughput than MIX at lower system loads between 60 to 86 active users. However at system loads from 87 to 110 active users, throughput of CLPSA exceeds the throughput of MIX. This is because at lower system loads, MIX due to its fair scheduling can allocate resources to users from all service types equally thus, increases system throughput. However at higher system loads, due to uneven distribution of different services MIX allocates the radio resource to the leftover users whatever channel conditions they have thus system throughput becomes low. The proposed CLPSA considers both the QoS provision and system throughput at the same time and tries to allocate resources adaptively, which results in maintaining a good system throughput at all system loads. Comparing with SWBS, the proposed CLPSA achieves higher system throughput at all system loads. As SWBS mainly considers QoS, it shows the lowest system throughput comparing with MIX and CLPSA. CLPSA maintains its good performance at all system loads. MIX shows good performance at lower system load, which becomes lower at higher system loads. This is because MIX cannot adapt itself to uneven distribution of services, which becomes more affective at higher system loads.

The proposed CLPSA is capable to maintain better performance in the changing network scenarios as compared to the state-of-the-art QoS aware MIX and SWBS algorithms. This is because CLPSA allocates resources to different services adaptively based on the achieved QoS, changing network conditions and variable system load, by using information on QoS measurements.

5.4.5 Performance Analysis of CLPSA

In the previous section, the performance of proposed CLPSA is compared with QoS aware SWBS and MIX algorithms under changing network scenarios. The proposed CLPSA is adaptive to the changing traffic patterns and variable number of active users. To validate it, this section gives the self-comparison of the proposed CLPSA under different scenarios.

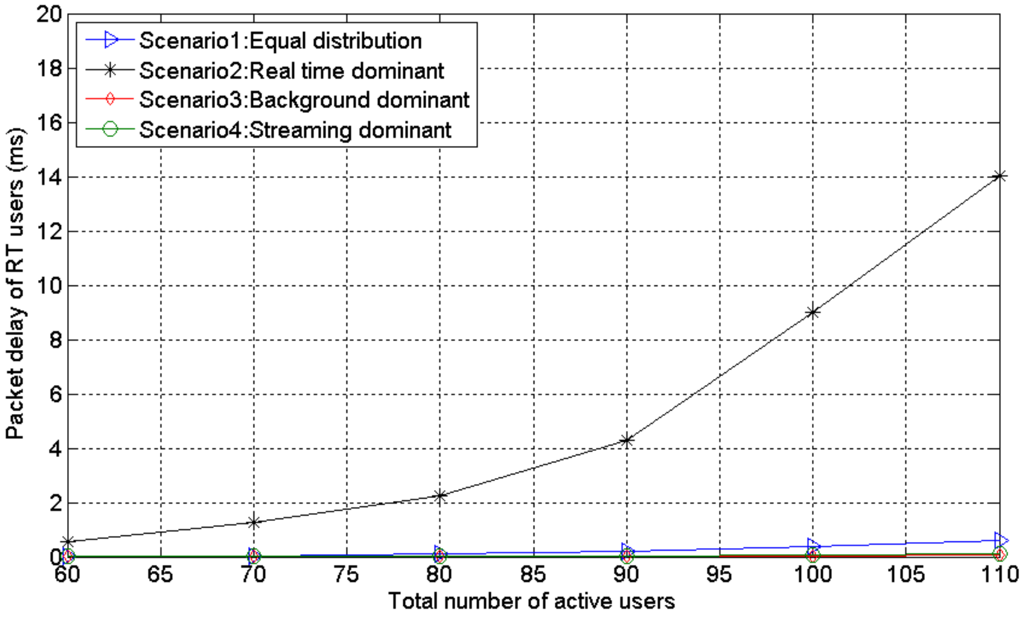


Figure 5-36 Average packet delay of real-time service

Figure 5-36 shows the average packet delay of real-time service for CLPSA under different system loads for all scenarios. It is very low in scenarios I, 3 and 4 in which real-time service is not dominant. However in scenario 2 where real-time service is dominant over all other services, the average packet delay increases with system load showing a value of 14 ms at the highest system load. However average packet delay remains under the delay upper bound which is 40 ms in LTE-A networks.

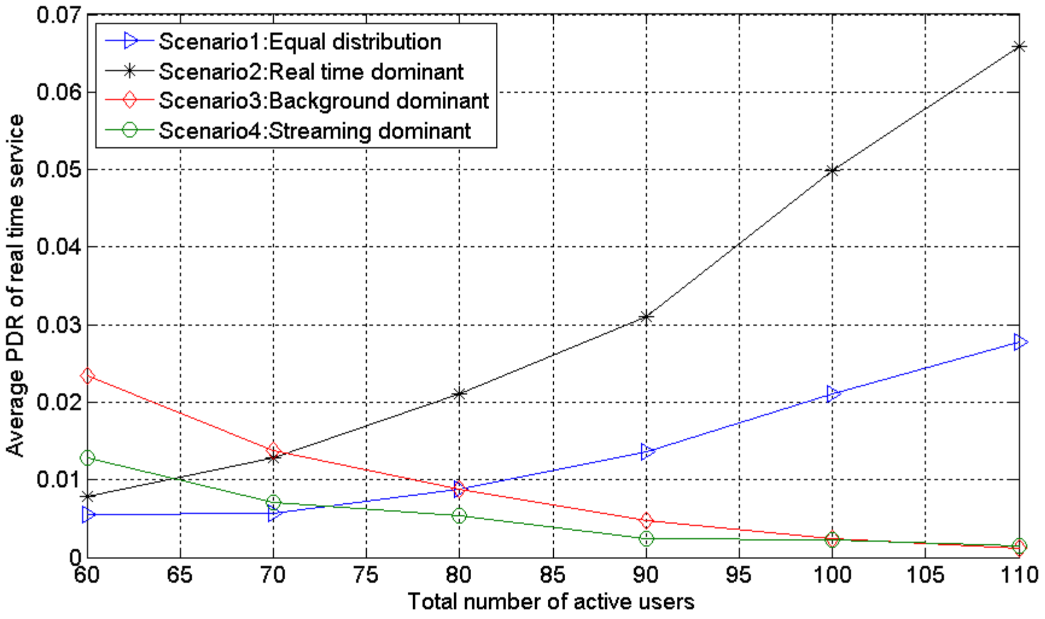


Figure 5-37 Average packet drop rate of real-time service

Average PDR of real-time service under different system loads for all scenarios is shown in Figure 5-37. In the First two scenarios, CLPSA shows the same trend of PDR and is increasing with the system load. This is because RT users are more in number in these scenarios as compared to the rest two scenarios. The total percentage of RT users is in these scenarios is equivalent to 50 % and 70 % of total system load, respectively. However in scenarios 3 and 4, average PDR is decreasing with system load because in these scenarios, RT users are only 20% of the total system load. However under all network scenarios, average PDR remains below the PDR threshold, which is 0.1 for LTE-A networks.

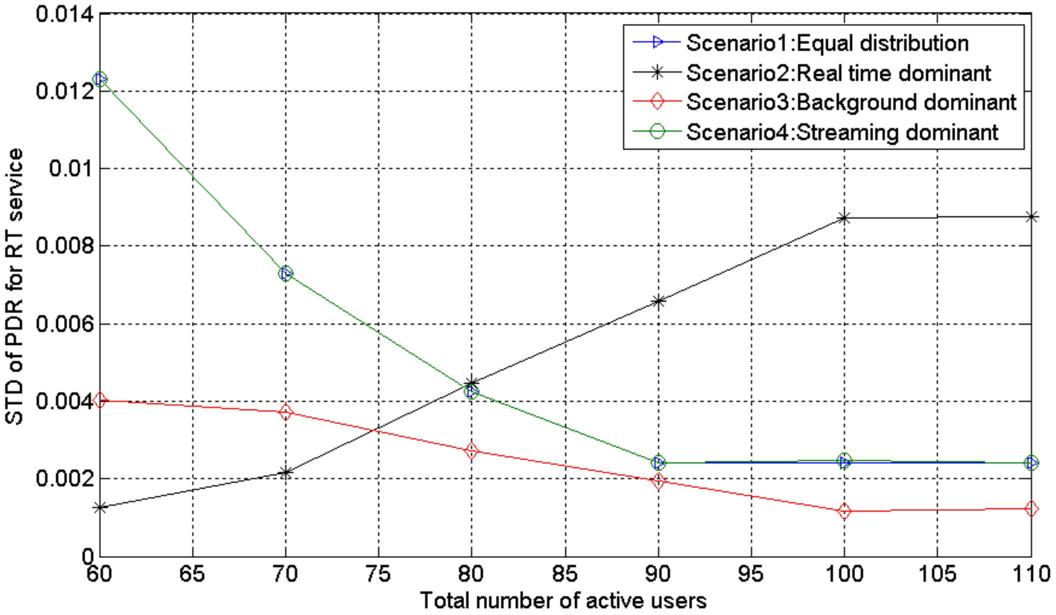


Figure 5-38 Standard deviation of average PDR distribution of RT users

Figure 5-38 shows STD of average PDR for all RT users with system loads, under different network scenarios. It remains under the PDR threshold (o.1) at all system loads and under all network scenarios. Scenario 2 however behaves differently by showing increasing STD values with system load, which becomes constant above 100 active users. This is because in scenario 4, real-time service is dominant over all other services. In scenario 1, 3 and 4, real-time service is not dominant, therefore STD values decrease with system load.

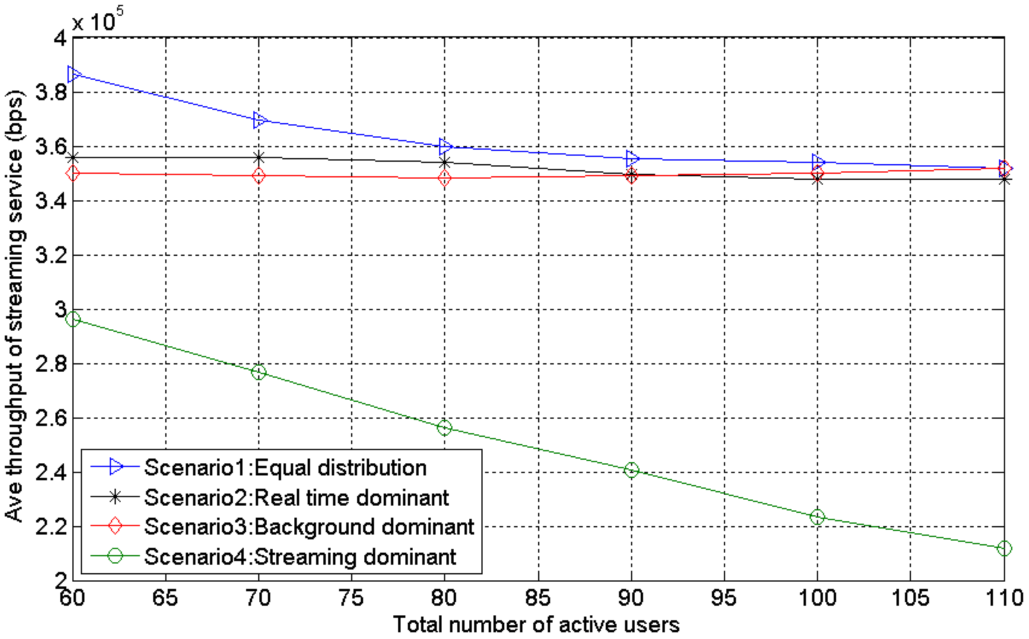


Figure 5-39 Average achieved throughput of streaming traffic

The average achieved throughput of streaming video service with system loads under different network scenarios is shown in Figure 5-39. The throughput requirement of streaming service is fulfilled in all scenarios except in scenario 4, in which streaming service is dominant over all other traffic types. The reason is the massive number of users demanding streaming service in this scenario. Under such condition, it is normal that all users cannot be guaranteed their throughput requirements.

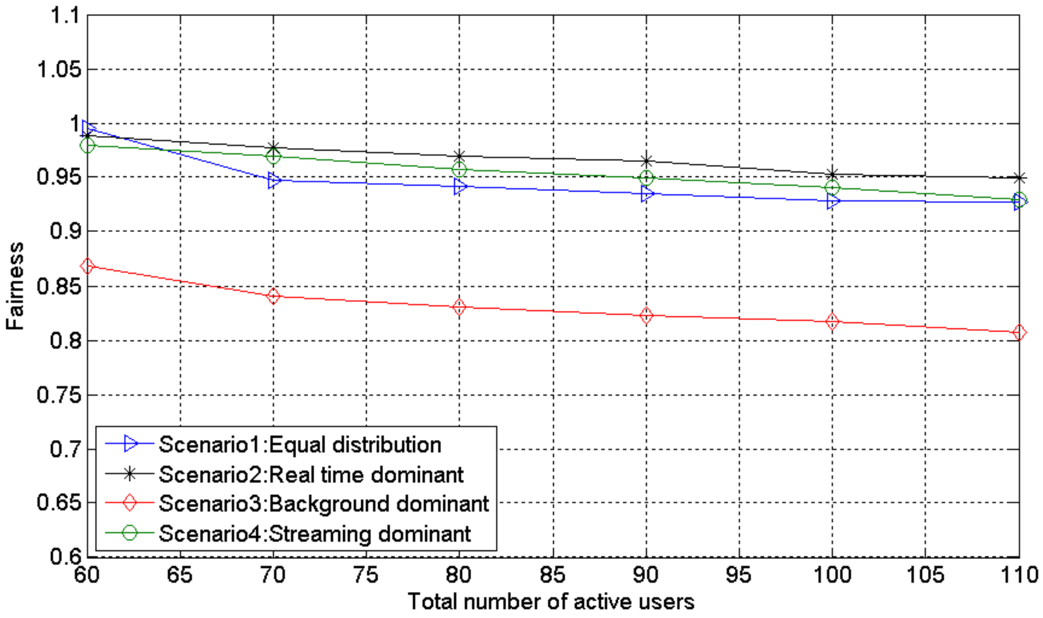


Figure 5-40 Fairness among all users

System-level performance is evaluated by system spectral efficiency and fairness among users of all traffic types.

The proposed CLPSA maintains the fairness at good level under all tested network scenarios as shown in Figure 5-40. Its value varies from 0.81 to 0.98 in different network scenarios, which is reasonably good fairness. In scenario 3 however, which considers equal number of RT and NRT users i.e. 20% each, CLPSA shows slightly lower fairness because firstly it allocates enough resources to real-time and streaming service to fulfil their delay and throughput requirements, respectively. Only then the remaining resources are allocated to background service, which is dominant i.e. 60% 0f total load in this scenario. In this way background service can get very low resources decreasing the overall fairness in the system.

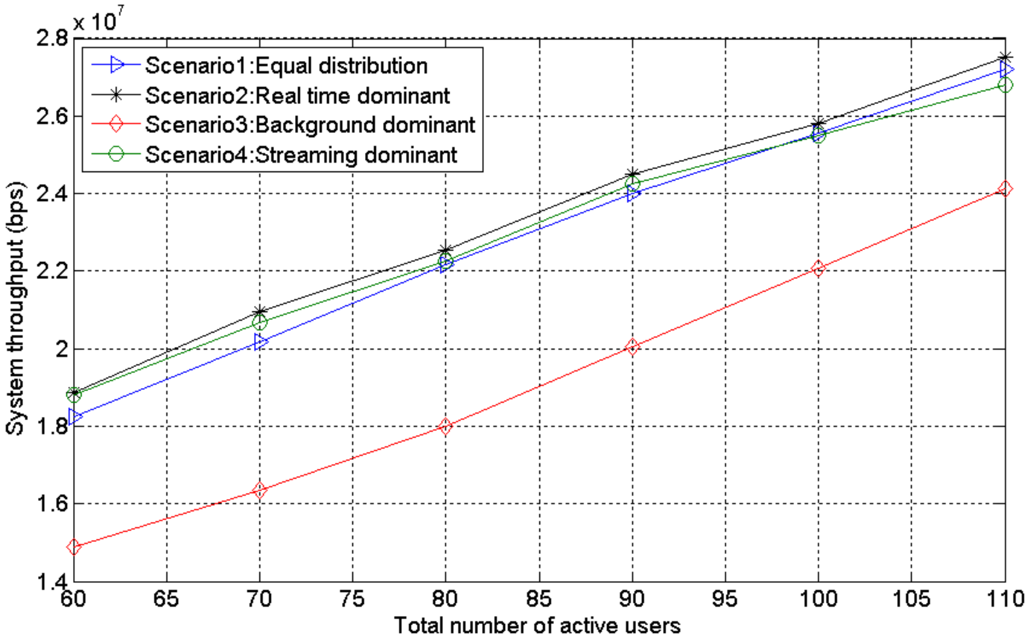


Figure 5-41 System throughput

System throughput with system loads is shown in Figure 5-41, which varies from 15 Mbps to 27.8 Mbps under different scenarios. In scenario 3 the proposed scheduling architecture shows lower system throughput as compared scenarios 1, 2 and 4. This is because the proposed CLPSA first considers the QoS requirements of real-time voice and streaming services. And users requiring these services may not have good channel condition, thus, decrease overall system throughput in this scenario.

The results show that the proposed CLPSA maintains good scheduling performance both at the user-level and at the system-level, under various network scenarios. It can work appropriately when the load of different services including real-time, non-real time and background is varied and also when the overall system load is varied from low to high.

5.5 Performance Evaluation of K-Mean Clustering Algorithm

To evaluate performance improvement made by the K-mean clustering algorithm, this section presents simulation results of proposed CLPSA with and without K-mean clustering algorithm. As K-mean clustering algorithm is incorporated to reduce the variation in the average PDR values of all RT users, the analyses presented in this section only include results on STD of average PDR values of all RT users and average PDR of real-time service. For these results the worst scenario in which system is highly loaded with real-time service (scenario 2), is considered. This is because K-mean algorithm is mainly concerned with real-time service so to best analyse its performance this scenario is selected.

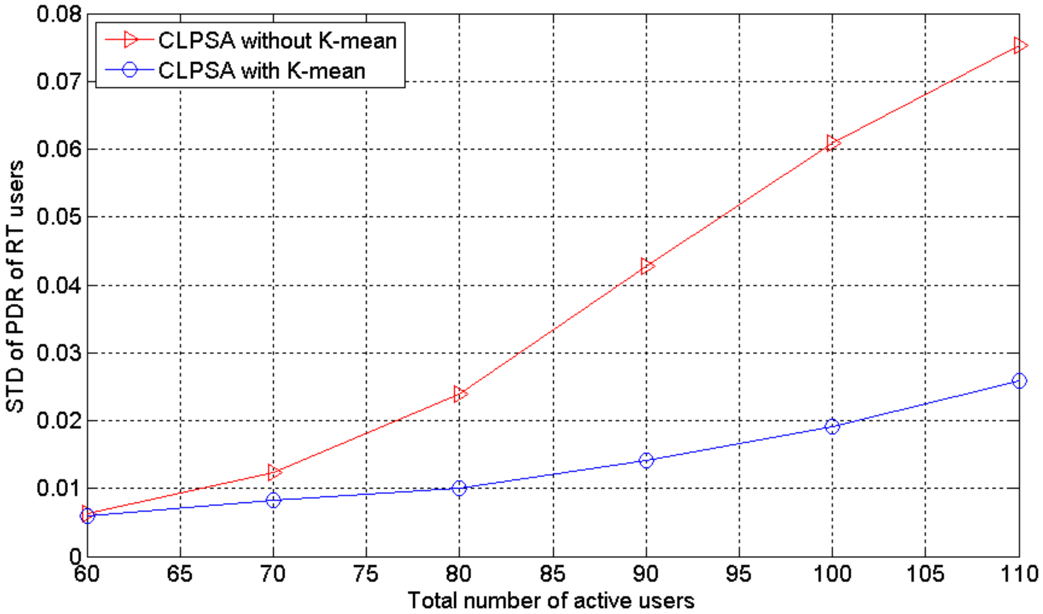


Figure 5-41 Standard deviation of average PDR distribution of RT users

Figure 5-41 shows the STD of average PDR of all RT users under different system loads without and with including K-mean clustering algorithm. With K-mean clustering algorithm, STD of real-time service becomes lower as the system load is increased. This is because K-mean clustering algorithm prioritises users with higher average PDR and becomes more effective when number of users increase as grouping effect becomes prominent. At lower system loads, due to smaller number of RT users, all users can get PRBs and meet their QoS demands, thus, clustering has little effect. However at higher system loads, when users are more in number, the competition to get PRBs becomes higher. In this scenario K-mean clustering can further arrange the priority of RT users based on their average PDR values and effectively lowers average PDR, which results in reduced STD of RT users.

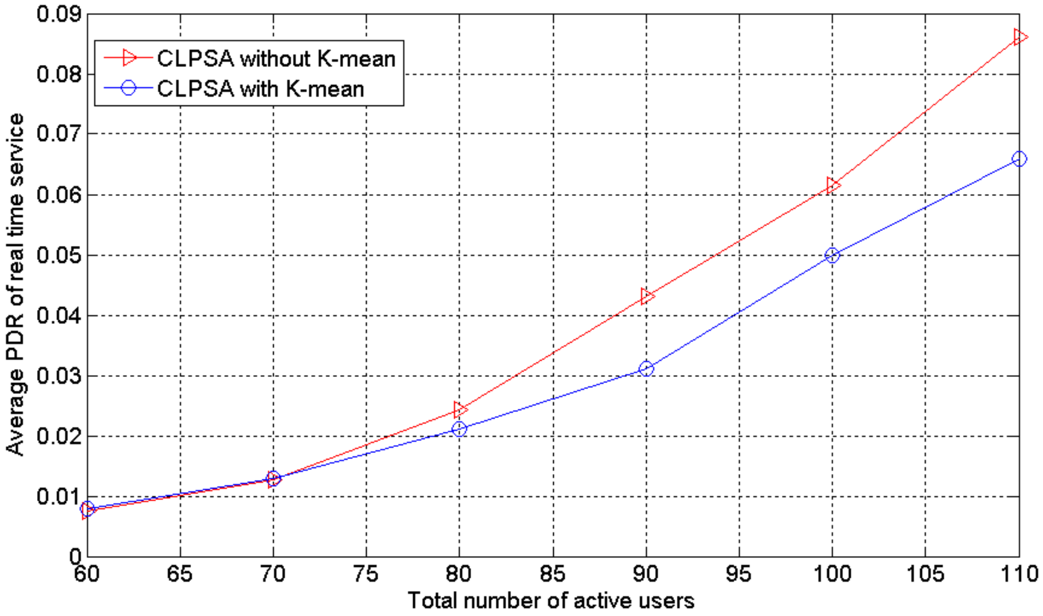


Figure 5-42 Average packet drop rate of real-time service

Figure 5-42 shows average PDR of real-time service under different system loads by using CLPSA with and without K-mean clustering algorithm. Average PDR of real-time service becomes lower at higher system loads when K-mean clustering algorithm is used. This is because K-mean clustering has already effectively reduced average PDR of individual RT users at highly loaded system, which ultimately reduces the average PDR of overall real-time service.

5.6 **Summary**

In this chapter simulation results and their analyses on all novel algorithms, proposed in this thesis, are presented. The performance of SSSA, Hebbian learning and K-mean Clustering algorithm are presented and compared with two QoS aware state-of-the-art algorithms. The overall performance of the packet scheduling architecture is tested under different network scenarios to exemplify the real systems. The results show an improved QoS to real-time service while fulfilling long-term rate requirements of non-real time service and maintaining a good trade-off between user-level and system-level performance.

Chapter 6 Conclusions and Future Work

6.1 Conclusions

A cross layer packet scheduling architecture (CLPSA) is proposed in this thesis for the DL transmission of LTE-A networks. In the CLPSA, two novel service specific queue sorting algorithms (SSSA) are proposed for the real-time voice service and non-real time streaming service, respectively, to improve user-level performance. An adaptive time domain scheduling algorithm (ATDSA) is proposed, which integrates Hebbian learning process and K-mean clustering algorithm to make scheduling decision by taking both system-level and user-level performance into account. The performance of the proposed SSSA and ATDSA including the effectiveness of K-mean clustering algorithm are investigated via system level simulation. The overall performance of CLPSA is analysed under different network scenarios including (i) equal distribution of real-time and non-real time services, (ii) real-time service dominant, (iii) background service dominant and (iv) streaming service dominant. The specific conclusions are:

- The proposed SSSA for prioritising real-time service in the *Traffic Differentiator* can effectively improve the QoS provision to RT users in terms of reduced average packet delay and average PDR in comparison to QoS aware SWBS algorithm. This performance improvement is more obvious at higher system load.

- The K-mean clustering algorithm integrated in the *TD Scheduler* can further reduce the average PDR of all RT users. In particular, it can effectively reduce the variation in average PDR of individual RT users, which leads to better fairness among RT users in terms of PDR. This phenomenon is most obvious in scenario (ii) where real-time service is dominant.

- The overall performance of the proposed CLPSA depends on the network scenarios:
    - In all four scenarios, CLPSA outperforms SWBS and MIX in terms of average packet delay of real-time service.
    - In scenario (ii) where the real-time service is dominant, CLPSA reduces the average packet delay and average PDR for real-time service in comparison with SWBS and MIX. At system-level, CLPSA outperforms SWBS; it achieves better fairness with a trade-off of a slight loss of system throughput as compared to MIX.
    - In scenario (iv) where the streaming service is dominant, CLPSA still achieves lower average packet delay of real-time service compared to SWBS and MIX. At system-level, compared to MIX, CLPSA shows similar performance; compared to SWBS, CLPSA achieves better fairness and higher system throughput with a trade-off of lower average throughput of steaming service.

6.2 Future Work

In this thesis, the main focus has been on scheduling in OFDMA-based LTE-A networks. One aspect that has been established is that right techniques can significantly improve the QoS provision to different services while maintaining a good trade-off between user-level and system-level performance.

However, there are other techniques that can improve the user-level performance particularly those near the cell edge. One such technique is relay-based OFDMA networks. One of the approaches for further work could be applying the principles of the methods of SSSA and ATDSA in relay-based networks. In particular, there has been no work focusing on learning environment data and taking scheduling decision based on it, in relay-based OFDMA networks so this is an area where there is significant potential.

References

[3GPP03} 3GPP Recommendation ITU-R M.1079-2, “Performance and quality of service requirements for mobile telecommunications 2000 (IMT 2000) access networks”, (1994-2000-2003).

[3GPP07] Recommendation ITU-R M.1822, “Framework for services supported by IMT”, 2007.

[3GPP09a] 3GPP TSG-RAN TS 36.300, “Evolved universal terrestrial radio access (E-UTRA) and evolved universal terrestrial radio access network (E-UTRANN)”, version 9.0.0, June 2009.

[3GPP09b] 3GPP TSG-RAN TR25.814, “Physical layer aspects for evolved UTRA”, version 7.0.0, June (2006-2009).

[3GPP09c] 3GPP Report ITU-R M.2135-1, “Guidelines for evolution of radio interface technologies for IM-Advanced”, Dec. 2009.

[3GPP11a] 3GPP TS 36.101, “Evolved universal terrestrial radio access (E-UTRA); user equipment (UE) radio transmission and reception”, version 10.4.0, Sep. 2011.

[3GPP11b] 3GPP Technical Specification group Radio Access Networks TR 36.913, “Requirements for Evolved UTRA (E-UTRA) and Evolved UTRAN (E-UTRAN)”, version 10.0.0 (2011-03).

[3GPP12a] 3GPP TR 36.912, “Feasibility study for further advancements for E-UTRA (LTE-Advanced)”, Release 9, 2009-2012.

[3GPP12b] 3GPP TR 36.912, “Feasibility study for further advancements for E-UTRA (LTE-Advanced)”, version 9.1.0 (2009-2012), pp. 42-43.

[3GPP12c] 3GPP TS 23.203, “Policy and charging control architecture”, Release 10, 2012.

[3GPPLTE12d] www.3gpp.org/Technologies/Keywoeds-Acronyms/LTE,

“The mobile broadband standards”, New opportunities for 3GPP in Rel-12, Barcelona 12th Dec. 2012.

[Adachi02] F. Adachi, “Evolution towards broadband wireless systems”, proceedings of the international symposium on wireless personal multimedia communications (WPMC), Vol. 1, Honolulu, Hawaii, October 2002, pp. 19-26.

[AG11] M. F. L. Abdullah and M. F. Ghanim, “An overview of CDMA technologies for mobile communications”, A journal of mobile communication, Vol.5, 2011, pp. 16-24.

[AKRSW01] M. Andrews, K. Kumaran, K. Ramanan, A. Stolyar and P. Whiting, "Providing Quality over a shared wireless link”, IEEE communications magazine, Vol. 39, issue 2, February 2001, pp. 150-154.

[AM08] M. Assaad, A. Mourad, ”New frequency-time scheduling algorithms for 3GPP/LTE-like OFDMA air interface in the downlink”, IEEE vehicular technology conference (VTC), 2008, pp. 1964-1969.

[APY95] J. B. Anderson, T. S. Rappaport and S. Yoshida, “Propagation measurements for wireless communication channels”, IEEE communication magazine, Vol. 33, issue, 1 Jan. 1995, pp. 42-44.

[AWEweb12] www-awe-communications.com/Propagation/Urban/COST/index.htm, “COST 231 Walfisch-Ikegami Model”, retrieved on Jan. 2012.

[Adas97] A. Adas, “Traffic models in broadband networks”, IEEE communication magazine, Vol. 35, issue 7, July 1997, pp. 82-89.

[CJAV05] W. Christian, O. Jan, G. E. Alexander, E. Von, ”Fairness and throughput analysis for generalized proportional fair scheduler in fourth generation wireless systems”, IEEE vehicular technology conference VTC-spring. 2005, pp. 1095-1098.

[CZWS10] J. Chen, L. Zhu, Q. Wu, Z. Shen, ”Multi-services supporting weighted power scheduling algorithm based on LWDF”, IEEE international conference on wireless communications and signal processing (WCSP) 2010, pp. 1-5.

[CD07] B. Chisung and Dong-Ho, “Fairness-aware adaptive resource allocation scheme in multihop OFDMA systems”, IEEE communication letters, Vol. 11, issue 2, Feb. 2007, pp. 134-136.

[Chandrasekaran06] B. Chandrasekaran, “Survey of network traffic models”, 2006, retrieved on Sep.2010 from: http://www.cse.wustl.edu/~jain/cse567-06/ftp/traffic_models3.pdf

[DPSB08] E. Dahlman, S. Parkvall, J. Skold and P. Bening, “3G Evolution; HSPA and LTE mobile broadband”, 2nd Ed., ISBN 978-0-12-374538-5, pp. 31-34.

[EFKMPTW06] H. Ekstrom, H. Furuskar, J. Karlsson, M. Meyer, S. Parkvall, J. Trsner, and M. Wahlqvist, ”Technical solution for 3G LTE”, IEEE communications magazine, Vol. 44, issue 3, March 2006, pp. 38-45.

[EHSB01] V. Erceg, K. V. S. Hari, M. S. Smith and D. S. Baum, “Channel models for fixed wireless applications”, contribution to IEEE 802.16.3, July 2001.

[FL96] B. H. Fleury and P. E. Leauthold, “Radiowave propagation in mobile communications: an overview of European research”, IEEE communication magazine, Vol. 34, issue 2, Feb. 1996, pp. 70-81.

[GBP08] I. Gutierrez, F. Bader, J. L. Pijoan, “Prioritization function for packet scheduling in OFDMA systems”, proceedings of annual international conference on wireless Nov. 2008, article no. 19.

[Goldsmith05] A. Goldsmith, “Wireless communications”, Cambridge University press 2005, ISBN 978-0-521-83716-3, pp. 10-12.

[GS97] A. J. Goldsmith and C. Soon-Ghee, “Variable-power MQAM for fading channels”, transaction on IEEE communications, Vol. 45, 1997, pp. 1218-1230.

[GPKM08] M. Guillaume, K. I. Pedersen, I. Z. Kovacs and P. E. Mogensen, “QoS oriented time and frequency domain packet schedulers for the UTRAN Long Term Evolution”, Proceedings of the IEEE vehicular technology conference (VTC), Singapore, May 2008, pp. 2532-2536.

[Hashmi93] H. Hashmi, “The indoor propagation channel”, proceedings of the IEEE, Vol. 81, issue 7 July 1993, pp. 943-968.

[HT09] H. Harri, A. Toskala, “LTE for UMTS OFDMA and SC-FDMA based radio access”, John Wiley & Sons, ISBN 978-0-470-99401-6 (H/B), 2009, pp. 5-6.

[Hebb49] D. O. Hebb (1949), “The organization of behaviour”, Wiley: New York, 1949.

[Haykin08] S. Haykin, “Neural networks and learning machines”, 3rd Ed., Vol. 10, ISBN-10: 0131471392, 2008.

[JN87] A. J. I. Jinning, and P. L. S. Nicholson, “Artificial intelligence in communication networks”, conference on computing systems and information technology, 1987.

[Jayakumari10] J. Jayakumari, “MIMO-OFDM for 4G wireless systems”, international journal of engineering science and technology, Vol. 2, issue 7, 2010.

[JL03] J. Jang and K. B. Lee, “Transmit power adaptation for multiuser OFDM systems”, IEEE journal on selected areas in communication, Vol. 21, issue 2, 2003, pp. 171-178.

[JWY05] A. Jamalipour, T. Wada and T. Yamazato, “A tutorial on multiple access technologies for beyond 3G mobile networks”, IEEE communications magazine, Vol. 43, issue 2, Feb. 2005, pp. 110-117.

[KSA08] C. B. Kian, A. Simon, D. Angela, “Joint time-frequency domain proportional fair scheduler with HARQ for 3GPP LTE Systems”, IEEE vehicular technology conference 2008, VTC-Fall, pp. 1-5.

[Kumar09] S. Kumar, “Techniques of efficient spectrum usage for next generation mobile communication networks-LTE and LTE-Advances study“, doctoral dissertation, University of Aalborg Denmark, June 2009.

[KALNN05] H. Kaaranen, A. Ahtiainen, S. Naghian, and V. Neime, Eds, “UMTS networks, architecture, mobility and services", 2nd Ed. John Wiley & Sons, ISBN 978-0-470-01103-4, 2005.

[KH10] F. Khozeimeh, S. Haykin, "Self-organizing dynamic spectrum management for cognitive networks", communication networks and services conference (CNSR), 2010, pp. 1-7.

[LL05] H. Liu and G. Li, "OFDM-based broadband wireless networks design and optimizations", John Willay & Sons, ISBN 978-0-471-72346-2, 2005.

[Molkdar91] D. Molkdar, "Review on radio propagation into and within buildings", IEEE Proceedings on microwave, antennas and propagation, Feb 1991, Vol. 138, issue 1, pp. 61-73.

[NH06] T. Nguyen, Y. Han,"A proportional fairness algorithm with QoS provision in downlink OFDMA systems", IEEE communication letters, Nov. 2006, Vol. 10, issue 11, pp. 760-762.

[PBA05] P. Parag, S. Bhashyam, and R. Aravind, "A subcarrier allocation algorithm for OFDMA using buffer and channel state information", IEEE vehicular technology conference, VTC-2005-Fall, 2005, pp. 622-625.

[PJNTTM03] K. Petter, P. Jani, K. Niko, R. Tapani, H. Tero, M. Martti, "Dynamic packet scheduling performance in UTRA Long Term Evaluation downlink", IEEE international symposium on wireless pervasive computing (ISWPC), 2008, pp. 308-313.

[JNTMM08] P. Jani, K. Niko, H. Tero, M. Martti and R. Mika, "Mixed traffic packet scheduling in UTRAN Long Term Evaluation downlink", IEEE international symposium on personal, indoor and mobile radio communications (PIMRC), 2008, pp. 1-5.

[Proakis01] J. G. Proakis, "Digital communications", ISBN 0-07-232111-3, McGraw-Hill, New York, 2001.

[PKVP06] S. Petridou, V. Koutsonikola, A. Vakali and G. Papadimitriou, “A divergence-oriented approach for web-users clustering”, proceedings of ICCSA, 2006, pp. 1229-1238.

[PPMRKM07] A. Pokhariyal, K. I. Pedersen, G. Monghal, I. Z. Kovac, C. Rosa, T. E. Kolding and P. E. Mogensen, ”HARQ aware frequency domain packet with different degrees of fairness for UTRAN LTE”, IEEE vehicular technology conference (VTC) 2007, pp. 2761-2765.

[PR80] A. Peled, A. Ruiz, ”Frequency domain data transmission using reduced computational complexity algorithms”, proceedings of ICASSP 1980, Vol. 5, pp. 964-967.

[PSPP07] S. G. Petridou, P. G. Sarigiannidis, Georgios I. Papadimitriou, A. S. Pomportsis, “Clustering based scheduling: A new approach to the design of scheduling algorithms for WDM star networks”, IEEE symposium on communications and vehicular technologies 2007, pp. 1-5.

[Rappaport02] T. S .Rappaport, “Wireless communications, principles and practice”, 2nd Ed., ISBN 0-13-042232-0, 2002, pp. 177-179.

[RM86] D. E. Rumelhart, J. L McClelland and PDP research group, “Parallel distributed processing: Explorations in the microstructure of cognition”, Cambridge Press, Vol. 1, ISBN 0-262-68053-X, 1986.

[RokeLTE13] Roke, “LTE MAC scheduler & radio bearer QoS”, retrieved on Jan. 2013 from, www.roke.co.uk/resources/white-papers/0485-LTE-Bearer-QoS.pdf

[Senarath07] G. Senarath, “Multi-Hop relay system evaluation methodology: Channel model and performance metric”, IEEE 802.16 broadband wireless access working group, Feb. 2007.

[SLC06] D. Soldani, M. Li, and R. Cuny, “QoS and QoE management in UMTS cellular networks”, John Willey & Sons Ltd, ISBN 978-0-470-01639-8, 2006.

[SR2000] A. L. Stolyar and K. Ramanan,” Largest weighted delay first scheduling: Large deviations and optimality”, The annals of applied probability, Vol. 11, No. 1, February 2000, pp. 1-48.

[Shannon48] C. E. Shannon, “A mathematical theory of communication”, Bell system technical journal, Vol. 27, July and Oct. 1948, pp. 379-423.

[SYLX09] J. Shen, N. Yi, A. Liu and H. Xiang, “Opportunistic scheduling for heterogeneous services in downlink OFDMA system”, IEEE international conference on communications and mobile computing, computer Society 2009, pp. 260-264.

[SCDT00] J. Srivastava, R. Cooley, M. Deshpande and P. N. Tan, “Web usage mining: discovery and application of usage patterns from web data”, SIGKDD Explorations, Vol. 1, No. 2, 2000, pp. 12-23.

[TCT10] T. U. Tsai, Y. L. Chung and Z. Tsai, “Communications and networking”, ISBN 978-953-114-5, published on Sep. 2010 under CCBY-NC-SA 3.0 licence, pp. 264-288.

[TV05] D. Tse, and P. Viswanath, “Fundamentals of wireless communication”, Cambridge University Press, 2005.

[WA93] W. Bechtel, and A. Abrahamsen, “Connectionism and the mind”, ISBN-10: 0631207120, Oxford, UK, Blackwell, 2002.

[WXZXY03] A. Wang , L. Xio, S. Zhou, X. Xu, Y. Yao, ”Dynamic resource management in the fourth generation wireless systems”, proceedings of communication technology, Vol. 2, ICCT international conference April 2003, pp. 1095-98.

[XC08] L. Xu, L. Cuthbert., “Improving fairness in relay-based access networks”, proceedings of international symposium on modelling, analysis and simulation of wireless and mobile systems (MSWIM) in ACM Nov. 2008, pp. 18-22.

[XCSCZ11] L. Xu, Y.Chen, J. Schormans, L. Cuthbert, T. Zhang, “User-vote assisted self-organizing load balancing for OFDMA cellular systems”, IEEE international symposium on personal, indoor and mobile radio communications (PIMRC) 2011, pp. 217-221.

[XW05] R. Xu and D. Wunsch,”Survey of Clustering Algorithms”, IEEE transaction on neural networks, Vol.16, no.3, May 2005, pp. 645-78.

[Yang05] H. Yang, “A road to future broadband wireless access: MIMO-OFDM-based air interface”, IEEE communication magazine, Vol. 43, no. 1, Jan. 2005, pp. 53-60.

[ZP02] G. Zhang, advisor prof. Panwar, “EL938 Report: Packet scheduling”, April 2002.