ISTANBUL TECHNICAL UNIVERSITY ★ GRADUATE SCHOOL OF SCIENCE ENGINEERING AND TECHNOLOGY

QoS-AWARE SCHEDULER FOR SELF-ORGANIZING LTE NETWORKS

M.Sc. THESIS

Beste AKKUZU

Department of Electronics & Communication Engineering

Telecomunication Engineering Programme

Thesis Advisor: Prof. Dr. Hakan Ali Çırpan

MAY 2014

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İSTANBUL TEKNİK ÜNİVERSİTESİ ★ FEN BİLİMLERİ ENSTİTÜSÜ

LTE AĞLAR İÇİN SERVİS KALİTESİNE DUYARLI KENDİNİ DÜZENLEYEN ÇİZELGELEYİCİ

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Beste AKKUZU
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MAYIS 2014

Beste Akkuzu , a **M.Sc.** student of ITU **Institute of Science and Technology of Telecomunication Engineering** student ID 504111308, successfully defended the **thesis** entitled "**QoS-AWARE SCHEDULER FOR SELF-ORGANIZING LTE NETWORKS**", which she prepared after fulfilling the requirements specified in the associated legislations, before the jury whose signatures are below.

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To my family,

FOREWORD

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Beste AKKUZU
Electronics and Communication Engineer

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ABBREVIATIONS

3GPP : Third Generation Partnership Project
ACK : Acknowledgement
ARQ : Automatic Repeat Request
AMC : Adaptive Modulation and Coding
CP : Cyclic Prefix
CRC : Cyclic Redundancy Check
CQI : Channel Quality Indicator
eNodeB : Evolved NodeB
EPC : Evolved Packet Core
EPS : Evolved Packet System
FTP : File Transfer Protocol
GSM : Global System for Mobile
GPRS : General Packet Radio Service
HARQ : Hybrid Automatic Repeat Request
Mbps : Megabits per second
MCS : Modulation and Coding Schemes
MIMO : Multiple Input Multiple Output
MLWDF : Modified Largest Weighted Delay First
NACK : Not Acknowledgement
NRT : Non Real Time
OFDM : Orthogonal Freqency Devision Multiplexing
OFDMA : Orthogonal Freqency Devision Multiple Access
PHY : Physical Layer
PDCP : Packet Data Convergence Protocol
PDCCH : Packet Data Control Channel
PDSCH : Packet Data Shared Channel
PF : Proportional Fair
QAM : Quadrature Amplitude Modulation
QoS : Quality of Service
RAN : Radio Access Network
RB : Resource Block
RLC : Radio Link Control
RR : Round Robin
RT : Real Time
SISO : Single Input Single Output
SINR : Signal with Interference to Noise Ratio
TCP : Transmission Control Protocol
TDD : Time Devision Duplex
TDM : Time Devision Multiplexing
TTI : Transmission Time Interval
UDP : User Datagram Protocol
UE : User Equipment

UMTS **:** Universal Mobile Telecomunication System
VoIP **:** Voice over Internet Protocol
VT-MLWDF : Virtual Token Modified Largest Delay First

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QoS-AWARE SCHEDULER FOR SELF-ORGANIZING LTE NETWORKS

SUMMARY

The convergence and interoperability of heterogeneous wired and wireless mobile networks became a reality with the recent advances in communication networks. Heterogeneous networks host multiple access technologies with varying capabilities such as coverage areas and Quality of Service (QoS) support. Higher rate wireless broadband communication such as Long Term Evolution (LTE) providing end-to-end IP based services will be the key enabler for achieving coherent heterogeneity in the context of future networks.

The widespread usage of smartphones reveals the necessity of higher data rates in mobile communication networks. Long Term Evolution (LTE), one of the 4th generation cellular technologies, has been developed as a next step of the 3G cellular systems to meet higher capacity and QoS requirements in mobile communication. Compared to former cellular technologes, LTE architecture is designed to implementation and management complexity, reduce the cost while providing higher data rates and spectral efficiency. In 3rd Generation Partnership Project (3GPP), LTE is introduced as a flexible radio interface which enables the digital convergence via carrying audio, video and data traffic end to end over IP packets.

Explosive growth of wireless data usage is expected to continue with the success of new and emerging mobile devices (e.g., smartphones). However, time varying link qualities and user mobility in wireless access networks require new mechanisms to support this traffic growth. Intelligent network management tools and more adaptive protocols providing self-optimization at different OSI layers are the key concepts to support efficient utilization of the limited resources in such complex networks. In this thesis, we propose a new LTE downlink scheduler with a reconfigurable resource prioritization mechanism to address these challenges and hence exploit new opportunities to improve the networking effectiveness and the user perceived QoS.

An LTE base station called evolved NodeB (eNodeB) includes the functionalities of both Base Station Controller (BSC) and Radio Network Controller (RNC), which were two distinct entities in the traditional cellular network architecture. The radio resource allocation task, which distributes available radio resources to UEs, is carried out within eNodeB. None of the existing standard scheduling algorithms such as Round Robin, MaxCQI and Proportional Fair takes channel qualities and traffic types into account at the same time when the available resources are allocated to UEs. For example, MaxCQI scheduler allocates Resource Blocks (RBs) to UEs solely using Channel Quality Indicator (CQI) information without taking the QoS requirements of UEs into account. These schedulers do not provide any instrument to influence the scheduling operation according to changing network conditions.

Self-Organizing Network (SON) is a type of network which optimizes the system parameters both at the deployment phase (planning stage) and also with respect to changing network conditions during run-time (e.g., changes in traffic demand, wireless link qualities, user locations, etc.). A SON feature for the LTE network can be realized in two steps: *(i)* extending the existing LTE scheduling algorithms to make them reconfigurable to varying network conditions, and *(ii)* reconfiguration of the LTE scheduling parameters when network conditions changes to reduce human involvement in network operations. In this thesis, a novel LTE downlink scheduling algorithm taking both channel qualities and traffic types of UEs into account is proposed to realize the first step of the SON feature. A reconfigurable parameter in the proposed scheduler determines the resource prioritization level for real-time services, and hence allows network administrators to more effectively control the resource allocation according to the QoS requirements of UEs and changing network dynamics.

Simulation experiments of the proposed scheduling algorithm are realized using the OMNeT++ simulation tool. The simulation results demonstrate that the tradeoff between the performance improvement of real-time flows and the performance degradation of nonreal-time flows can be controlled by tuning the reconfigurable traffic prioritization parameter of the proposed scheduling algorithm. In future work the proposed algorithm will be extended to realize fully automated SON concept.

LTE AĞLAR İÇİN SERVİS KALİTESİNE DUYARLI KENDİNİ DÜZENLEYEN ÇİZELGELEYİCİ

ÖZET

Son zamanlarda akıllı mobil cihazların kullanımının yaygınlaşması, mobil iletişimde yüksek veri hızına olan ihtiyacı ortaya çıkarmıştır. Bu ihtiyacın karşılanması amacıyla 4. nesil iletişim teknolojilerinden Long Term Evalution (LTE), 3G hücresel sistemlerin bir sonraki aşaması olarak ve hücresel mobil iletişimin daha yüksek performanslarla yapılabilmesi için geliştirilmiştir. Önceki hücresel sistemlerle karşılaştırıldığında, LTE'nin yeni radyo-ağ mimarisi işlemleri basitleştirmek, maliyeti azaltmak ve hücresel haberleşmenin daha yüksek hızlarda ve spektral verimlilikle yapılabilmesi için tasarlanmıştır. 3rd Generation Partnership Project (3GPP) tarafından esnek radyo arayüzü olarak tanımlanan LTE'nin sunduğu yüksek kapasiteler, yüksek veri hızı, yüksek spektral verimilik, düşük gecikme süreleri sayesinde; ses, görüntü ve veri trafiği uçtan uca IP paketler ile taşınarak dijital yakınsama gerçekleştirilir.

Kablolu ve kablosuz çok türel ağlarda, yakınsama ve birlikte çalışabilirlik haberleşme ağlarındaki son gelişmelerle beraber gerçeklik kazanmıştır. Çok türel ağlar, kapsama alanları, QoS (Quality of Service) desteği gibi farklı yetenekleri sayesinde çoklu erişim teknolojilerine hizmet eder. Geniş bantlı kablosuz iletişimde yüksek veri hızı sağlayan LTE, uçtan uca IP tabanlı servisleriyle gelecekteki çok türlü mobil ağların uyumunu gerçekleştirecek anahtar teknoloji konumundadır. Bu sayede uçtan uca hizmet vermesiyle beraber kesintisiz haberlişmeyi de destekler niteliktedir.

Akıllı mobil cihazların (örneğin, akıllı telefonlar) yaygınlaşması sonucu mobil iletişimde yüksek veri hızına olan ihtiyacın artarak devam etmesi beklenmektedir. Kablosuz erişim ağlarında hat kalitesinin dinamik olarak değişmesi ve kullanıcıların hareketli olması nedeniyle bu talep artışını karşılamak için yeni mekanizmaların geliştirilmesi gerekmektedir. Akıllı ağ yönetim araçları ve adaptif protokoller farklı OSI katmanlarında insan eli değmeden optimizasyon sağlayarak sınırlı kaynakların verimli kullanımını destekler niteliktedir. Bu bağlamda, Self Organizing Network (SON) LTE'deki önemli kavramlardan biri olarak öne çıkmaktadır. Ağ birimlerinden alınan ölçümler kullanılarak LTE parametrelerinde dinamik ayarlamalar yapılarak değişen ağ durumuna göre performans en yüksek hale getirilmeye çalışılır ve böylece ağ operatörleri için maliyet etkin altyapıların oluşturulması sağlanır. SON'nin bu özellikleri ancak, *(i)* var olan LTE çizelgeleme algoritmalarının değişkenlik gösteren ağ durumlarına göre kendi kendini düzenleyebilir halde geliştirlmesiyle, *(ii)* LTE çizelgeleme parametrelerinin ağ koşullarına göre yeniden yapılandırılmasıyla gerçekleştirilebilir.

Bu çalışmada, ağların verimli kullanımını sağlayan ve kullanıcıların aldığı servis kalitesine duyarlı yeni bir çizelgeleyici önerilmektedir. Önerilen çizelgeleyicinin

ayarlanabilir şekilde kaynak önceliklendirmesi yapan mekanizmaya sahip olması, SON kapsamında belirtilen sorunları adresleyerek yönetilebilir ağları ortaya çıkarmaktadır. Tezde öne sürülen çizelgeleyici algoritmalarının hem kullanıcıların kanal kalitelerine hem de trafik türlerini gözönüne alarak çalışması SON'nın ilk şartını gerçeklemektedir. Ayrıca yeniden yapılandırılabilir LTE parametresinin var oluşu ve bu parametrenin gerçek zamanlı kullanıcıların kaynaklarının önceliklendirilmesinde kullanılması ikinci koşulu da sağlamaktadır. Böylece ağ yöneticisine değişen ağ dinamiklerine göre kullanıcıların QoS gerekliliklerini sağlamak için var olan kaynakları kullanıcılara daha etkin dağıtabilmesi için daha güçlü bir kontrol verilmiş olur.

Daha önceki teknolojilerden farklı olarak LTE teknolojisinde kaynakları kontrol edecek RNC (Radio Resource Controller) veya BSC (Base Station Controller) gibi merkezi kontrol birimleri yoktur. Bu yüzden kullanıcı kabul denetimi, MAC (Medium Access Controller) çizelgeleme gibi radyo kaynaklarının yönetimi baz istasyonu eNodeB (evolved NodeB) üzerinde gerçekleşir. Kısacası, baz istasyonu var olan fiziksel katman kaynaklarını kullanıcılara tahsis eder. Bu fiziksel kaynakların kullanıcılara dağıtılması işlemi çizelgeleme işlemidir ve baz istasyonu tarafindan gerçekleştirilir.

Adaptif Kodlama ve Modülasyon (AMC) kanal şartlarına bağlı olarak modulasyon seviyesi ve kodlama oranının ayarlanmasını sağlar. Örneğin, kanal kalitesi iyi olan bir kullanıcı 64QAM ile kodlanırken kanal kalitesi düşük bir kanaldan gelen kullanıcı 16QAM kodlamaya tabi olur. Böylece kanal kalitesine bağlı çizelgeleme işlemi yapılabilir ve radyo kaynakları daha verimli kullanılabilir. Baz istasyonu ise bağlı olan kullanıcılardan periyodik olarak kullanıcıların kanal kalite bilgisini (Channel Quality Information) alır. Kullanıcının kanal kalitesi arttıkça CQI değeri de artar.

LTE'de paket iletimi radyo çerçeveleriyle sağlanır. Her bir radyo çerçeve 10 tane alt çerçeveye bölünmüştür. Her bir alt çerçeve 1 ms uzunluğundadır. Dolayısıyla bir çerçeve 10 ms uzunluğa sahiptir. Alt çerçeveler ise 0.5 ms uzunluklu iki zaman dilimine ayrılmıştır ve her bir zaman dilimi 6 veya 7 OFDM sembölü içerir. Sembol sayısı normal veya uzun çevirmeli önek kullanılıp kullanılmamasıyla alakalıdır. 0.5 ms uzunluklu zaman dilimleri aynı zamanda birer kaynak bloğuna karşılık gelmektedir. Yani bir zaman dilimi aynı zamanda bir kaynağı RB (Resource Block) ifade eder. Bir RB'nin bant genişliği 180 kHz'dir. Çizelgeleyici her 1ms'lik alt çerçeve için hangi kullanıcının iletimine izin verileceğini, hangi frekans kaynaklarının kullanılacağını ve hangi hızda iletim yapılacağını belirler. Eğer sönümlü bir kanal söz konusu ise kanal kalitesi zamana ve frekansa göre değişkenlik gösterebilir. Özellikle bu tarz durumlarda çizelgeleyicinin önemi ortaya çıkar.

LTE'de 1.4 MHz ile 20 MHz arasında değişen bant genişliği kullanılır. LTE'nin bu özelliği asimetrik spektrum kullanımı mümkün kılar, yani kullanıcının durumuna göre farklı bant genişlikleri tercih edilebilir. Bant genişliğine göre RB sayısı da değişir. Örneğin 1.4 MHz seçildiğinde 6 RB sunulurken, 20 MHz'de bu sayı 100 RB'a karşılık gelir. Böylece daha fazla kaynak baz istasyonundan kullanıcıya tahsis edilmiş olur. Ne var ki sistemin güç tüketimi, ağ şartları veyahut kullanıcı sayısına göre bu kaynaklar ayarlanabilir. Her bir kaynak ise 12 veya 14 OFDM alt taşıyıcılardan oluşur ve her bir alt taşıyıcının bant genişliği ise 15 kHz'dir. LTE'de kaynak iletimi aynı frekans bandında gerçekleşebileceği gibi farklı frekans bandında da gerçekleşebilir. OFDMA ile verinin çok sayıda paralel, dar-bant alt taşıyıcı ile iletilmesi sağlanır. OFDMA iletimi sayesinde alıcı kısmındaki işlemler kolaylaşır, terminal kısmı giderleri ve güç tüketimi azalır, ayrıca genişbant ve çoklu anten kullanımını desteklemesi iletimde önemli avantajlar sağlar. Sunulan çoklu antenler ile yüksek data hızlarına ulaşılabilir. Tekli anten kullanımında SISO (Single Input Single Output), maksimum hücre verimliliği 100Mbps olarak indirme yönünde gerçekleşir. Bu hızlar çoklu antenlerle MIMO (Multiple Input Multiple Output) 2x2 ile 2 katına, MIMO 4x4 antenlerle ile 4 katına kadar çıkartılabilir.

Bugüne kadar sistemdeki bu radyo kaynaklarının daha etkili bir biçimde kullanılabilmesi için çeşitli yöntemlere dayanan farklı çizelgeleme algoritmaları geliştirilmiştir. Bunlardan yaygın olarak kullanılan çizelgeleme algoritmaları; kanal kalitesi yüksek kullanıcıya öncelik veren MaxC/I, kaynakların kullanıcılar arasında adil kullanımını dikkate alan Round Robin, adil kullanıma dikkat etmekle beraber kullanıcılara farklı ağırlıkta önem veren Propotional Fair algoritmalarıdır. MaxC/I algoritması kanal kalitesi yüksek olan kullanıcıya en çok kaynağı sağladığından, kanal kalitesi iyi olan kullanıcılar yüksek veri hızlarına kolayca ulaşırlarken, kanal kalitesi düşük olan kullanıcılar en dezavantajlı konuma düşerler. Round Robin algoritmasında ise kaynaklar kullanıcılara sırasıyla dağıtılır. Kullanıcılar arasında adil bir kullanım sağlamasına rağmen kanal kalitesi düşük olan kullanıcıya harcanan kaynaklar ile algoritma verimsiz hale gelir. Propotional Fair algoritmasında ise kullanıcılara farklı ağırlıkta önem verilerek kanallar kullanıcılar arasında paylaştırılır. Var olan algoritmalardan farklı olarak bu çalışmada önerdiğimiz algoritma; gerçek zamanlı kullanıcıların servis kalitesini desteklemek amacıyla, MaxC/I algoritmasının trafik türüne göre de kaynak tahsis yapacak şekilde ayarlanmış ayrıca önerilen parametre ile kullanıcılar arasındaki adil kullanım oranı ve hızların ayarlanabilirliği sağlanmiştir.

Literatürde özellikle ağ iletişimi alanında kabul görmüş, açık kaynak kodlu OMNeT++ kullanılarak yapılan simulasyonlarla gerçek zamanlı ve gerçek olmayan zamanlı kullanıcılara önerilen çizelgeleyici algoritmasıyla kaynak tahsisi yapılmıştır. OMNeT++'daki MaxC/I çizelgeleme algoritması kullanıcıları kanal kalitesine göre bir sıraya dizer. Kaynaklar kullanıcılara tahsis edilirken bu sıranın en önündeki kullanıcıdan başlanarak "Round Robin" mekanizmasına göre dağıtılır. Yani kanal kalitesi en iyi olan kullanıcı sıranın en önünde yer alır ve MaxC/I algoritması için en avantajlı kullanıcıdır.

Önerdiğimiz çizelgeleyici için üç farklı algoritma tasarlanmıştır. İlk algoritma trafik türüne göre ayırdığı kullanıclardan gerçek zamanlı olanlarını tamamen önceliklendirerek çalışır. İkincil önerilen algoritma ise gerçek zamanlı kullanıcıları önceliklendirdiği gibi; kanal kalitesi yüksek, gerçek olmayan zamanlı kullanıcıların da kaynaklardan faydalanmasına olanak tanır. Bu algoritma ile beraber gelen yeniden yapılandırılabilir parametre gerçek zamanlı trafiğin gecikme değerindeki iyileştirme ile taşınan gerçek zamanlı olmayan veri miktarındaki azalma arasındaki değiş tokuşun ağ yöneticisi tarafından ayarlanabilmesini sağlar. Son olarak geliştirdiğimiz algoritma ise artan VoIP ve video trafiğinin servis kalitesini destekleyerek gerçek zamanlı trafik kullanıcılarına daha iyi hizmet verecek şekilde tasarlanmıştır. Simulasyon sonuçları MaxC/I ve PF algoritmalarına göre karşılastırıldığında adil kullanım oranları ve verimlilik arasındaki değiş tokuşu sağlayabildiğini, aynı zamanda gereken düşük gecikme sürelerini desteklediğini göstermektedir.

1. INTRODUCTION

An explosive growth in the number of mobile users necessitates higher capacities in the cellular network infrastructure. Emerging real-time services such as VoIP and video streaming, online gaming have stringent delay requirements. 3GPP introduces LTE which has been developed to fulfill the requirements of 4G cellular networks such as higher data rates, lower latency, for real-time services and lower cost of operation with its all-IP based architecture [1]. Therefore, many academic and industrial studies have been carried out to realize high performing LTE network using innovative solutions.

Compared to previous networks, LTE introduces as a flexible radio interface to provide interopabilitiy of heterogenous networks [2], flexible bandwidth usage and support end to end IP based services. Therefore, it will be key enabler to ensure Quality of Service (QoS) perceived by user with higher data rates (i.e., the allowed peak data rates, higher spectral efficiency and lower delay. To meet these objectives, Medium Access Control (MAC) layer functions such as resource management, Channel Quality Indicator (CQI) reporting and link adaptation through Adaptive Modulation and Coding (AMC) to have a key role. In this regard, efficient resource allocation according to QoS requirements of users are realized within evolved NodeB (eNodeB) which includes the functionalities of both Base Station Controller (BSC) and Radio Network Controller (RNC). These were two used to be performed distinct entities in the traditional cellular network architecture.

Physical radio resources of eNodeB can be seen as frequency time grid. In the frequency domain, LTE network exploits Orthogonal Frequency Division Multiple Access (OFDMA) at the downlink which uses narrow-band subcarriers distributed in the entire spectrum. In the time domain, time is divided into frames and sub-frames such that duration of each subframe is 1ms. In LTE, a resource block (RB), which is the minimum resource unit assigned to UE, corresponds to the bandwidth of 180kHz in the frequency domain and 0.5ms in the time domain. In this context, the scheduler located at the base station plays a key role by assigning the resource block to the mobile terminals. The scheduler determines for each sub-frame which user(s) are permitted to transmit and on what frequency resources the transmission will take place. According to the operating bandwidth from 1.4MHz to 20MHz, number of available RB is changes from 6 to 100, respectively.

LTE MAC scheduler plays a crucial role to support efficient utilization of the limited radio resources amoung users in such complex networks. In this thesis, we propose a new LTE downlink scheduler with a reconfigurable resource prioritization mechanism to exploit new opportunities to improve the networking effectiveness and the user perceived QoS. In next generation wireless networks will use self-organizing features to achieve two main objectives; the first is to reduce human involvement in network operational tasks and second one is to improve network coverage and service quality by automatically adapting its resource allocation decision when the network condition change. Our proposed scheduling algorithms are designed to achieve Self Organizing Network (SON) concept [3], by taking traffic type into account to prioritize resources for real-time users and providing a reconfigurable parameter to minimize the human involvement for network operations.

In this thesis, simulation experiment ar performed by using OMNeT++ simulation tool which has been heavily used by the researchers in the area of communication networks. It is an open source tool in which we analyze and simulate proposed scheduling algorithms for LTE access network. OMNeT++ based simulation results demonstrate that the tradeoff between the performance improvement of real-time flows and the performance degradation of nonreal-time flows can be controlled by tuning the reconfigurable traffic prioritization parameter of the proposed scheduling algorithms.

This section gives an overview of the LTE network architecture and our novel work on scheduling algorithms. Next, the purpose of the thesis, literature review and the hypothesis of the relevant study is presented.

1.1 Purpose of Thesis

In this thesis, novel LTE downlink scheduling algorithms, taking both channel qualities and traffic types of UEs into account, are proposed to realize the SON concept. For this purpose, the existing LTE scheduling are extended to make them reconfigurable to varying network conditions. Make them reconfigurable to provide dynamic adaptation when the network conditions vary. A reconfigurable parameter in the proposed schedulers determines the resource prioritization level for real-time services, and hence allows network administrators to more effectively control the resource allocation according to the QoS requirements of UEs and changing network dynamics. OMNeT++ will be used to study the performance of scheduling algorithm.

1.2 Literature Review

Resource allocation problem has been extensively studies in the context of varies network to maximize the network performance and fulfill the QoS requirements of user. In LTE, a downlink scheduling is the process used for resource allocation at the Medium Access Control layer and underlying technology is Orthogonal Frequency Division Multiple Access. It is essential to design efficient resource scheduling algorithms to improve the performance of data services and end user experiences. In this context, scheduling algorthms are evaluated according to several performance metrics, such as packet loss, throughput, fairness, spectral efficinecy, delay, power reduction, etc.

Generally, scheduling can be divided into two classes: channel-independent scheduling and channel-dependent scheduling [26]. Channel-independent scheduling does not take channel conditions into account. On the contrary, channel-dependent scheduling is performed by allocating resources based on channel conditions. It deals with how to share available radio resources between different terminals to achieve efficient resource utilization as possible. This implies to minimize the amount of resources needed per user while still satisfying quality of service requirements. Thus, as many users as possible in the system can use the resources.

Furthermore scheduling can be classified based on application scenarios, since the performance of scheduling algorithms highly relies on the type of incoming flows. In order to achieve application specific performance, it is important to select suitable algorithms according to traffic types of flows which can be divided into Real-Time and Non-Real-Time by considering LTE services, including such as voice, data, and video [23].

The most popular scheduling algorithms for nonreal-time flows in LTE networks are Round Robin [25], MaxC/I [26] and Proportional Fair [29]. The details of algorithms are in given in Chapter 3. Recent studies on LTE scheduling are summurized below.

Ref [4] extend, virtual token modified largest weighted delay first (VT-MLWDF) algorithm and token modified largest weighted delay first algorithm (MLWDF) to achieve inter-class fairness for resource allocation. VT-MLWDF focuses on real-time (RT) services whereas a minimum throughput is provided for nonreal-time (NRT) services. Constitutively, this scheduler is based on the queue size of each RT flow's buffer to allocate most of the resources to video streaming services. On the other hand, MLWDF algorithm mainly relies on the delay of the packets for RT, while NRT services are scheduled using standart PF scheduler. Their algorithm consider queue size of each class and end ot end packet delay of each class where such that video and VoIP traffic for RT class and constant bit rate traffic for NRT class are used. It has better performance with respect to non-classification weighted algorithms in terms of packet loss ratio, average throughput, fairness and system spectral density. The distribution of the users, in terms of traffic classes is essential to provide an adequate QoS, however they do not consider using various services under different load situations.

Ref [5], the authors propose two new algorithms based on PF metric and design to enhance delay performance of RT users. One of the algorithms considers delay performance and gives almost all resources to RT users. In the second approach, they prioritize all packets of real-time traffic that are lower than delay-constraint, however priority factor is based on simulation results and do not represent optimal solution.

In [6], improved resource scheduling algorithm is proposed by extending the existing semi-contionuous scheduling algorithm for VoIP data packets. Their scheme maps the TCP packets to higher priority to decrease the occurrence of discarded ACK packets and avoid the frequent use of TCP congestion control mechanism that would increase the round trip time latencies and reduce system throughput. Nevertheless, their mapping mechanism working on PDCP layer in LTE and only consider the TCP acknowledgement packet to map idle logical channel with higher priority.

In the paper [7], their target is developing a new scheduling strategy for LTE-Advanced in relay-enhanced networks. Their offer is on cells enhanced by outdoor in-band relays. The QoS-aware metric they introduce is the multiplication form of the end to end delay, service priority, rate and proportional fair. Their controlling mechanisim works between bachaul and direct link of the user equipments to relay nodes and eNodeBs. Rearranging quality channel indicators to the users according to their traffic types such as video streaming, VoIP and NRT traffic, they allocate resources to relay node and eNodeB for UEs. Nonetheless, their simulations are not realized for different load scenarios and not take into account the real channel indicator parameters.

Ref [8] proposes a new scheduling algorithm which considers scalable video coding. Layered coding schema of video streaming provides better service qualities under different network loads. In [9], their proposed algorithm is based on proportional fair packet scheduling which categorizes real-time and nonreal-time services to maintain the fairness of all services and avoid the latency or starvation. However, their proposed algorithm has not been evaluated in any simulator. Ref [10] presents a downlink scheduling strategy for video transmission relying on cooperative game theory.

1.3 Hypothesis

Unlike previous works on resource allocation; in this thesis, we propose novel reconfigurable scheduler algorithms by considering the channel qualities and traffic types at the same time to provide certain QoS guaranteed for real-time traffic and manage overall network effectiveness. For example, the MaxC/I scheduler decides allocation of Resource Blocks (RBs) according to Channel Quality Indicator (CQI) parameters and hence it provides higher throughputs while there is no service guarantee for real-time traffic [11].

Our proposed scheduler takes both channel qualities and the traffic types into consideration and gives an opportunity to the administrator to control the tradeoff between the performance of real-time and non-real-time traffic. First, incoming traffic is classified into real-time and nonreal-time flows which are inserted into a priority list according to traffic types and channel qualities. The scheduler assigns available resource blocks to user equipments (UEs) using the order in the list. A reconfigurable parameter in the proposed scheduler determines the resource prioritization level for real-time services, and hence allows network administrators to more effectively control the resource allocation according to the QoS requirements of UEs and changing network dynamics.

We conjecture that our proposed scheduling can effectively used to provide efficient utilization of limited radio resources in time varying network condition.

2. BACKGROUND INFORMATION

2.1 Long Term Evolution (LTE)

3GPP introduces LTE by “LTE Release 8”, called also the E-UTRAN (Evolved Universal Terrestrial Access Network) as the access part of the Evolved Packet System (EPS). LTE Release 8 is the first LTE standart released in December 2008 [12]. Release 9, Release 10, and Release 11 versions of LTE are evolution step taken over LTE release 8. After the Release 10 organization give a name of the developed technology as LTE-Advanced.

The main reason behind the world’s switching to LTE technology is its promise of high end-user data transfer rates, low delay and latency, and high spectral efficiency. With LTE, people can get higher data transfer rates of up to 300 Mbps, with latency less than 5 milli-seconds [13]. The most powerful multicarrier transmission technology called Orthogonal Frequency Division Multiplexing (OFDM) with Multiple Input Multiple Output (MIMO) scheme has made this high data transfer rate possible in LTE. Figure 2.1 illustrates the network evolution from second generation Global System for Mobile (GSM), intermediate formation General Packet Radio Service (GPRS), third generation Universial Mobile Communication System (UMTS) to Long Term Evolution (LTE).

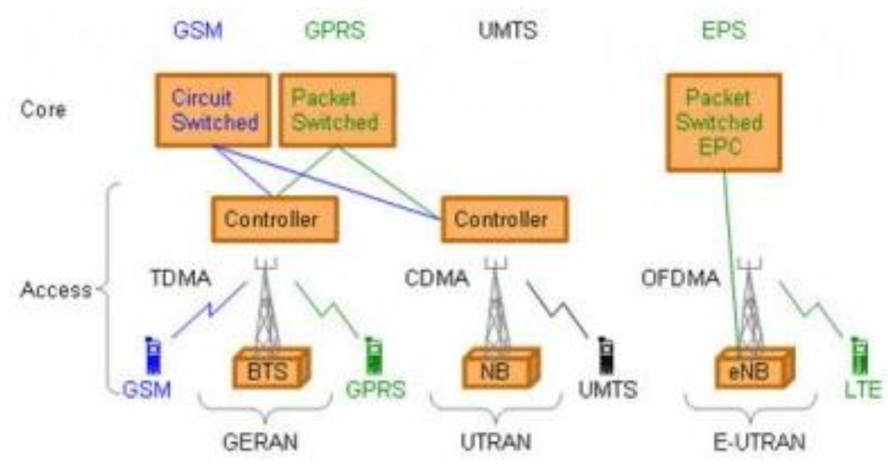

Figure 2.1 : Cellular Network Evolution from GSM to LTE[36].

From the Figure 2.1, it can be seen that GSM is connected circuit switch box with blue link. It carries real-time services in traditional circuit switched way. The data services are only available over a circuit switched modem connection, with very low data rates. The first step towards an IP based packet switched solution is demonstrated by green link that was taken with the evolution of GSM to GPRS, using the same air interface and access method, TDMA (Time Division Multiple Access).

Later, the new access technology is developed, WCDMA (Wideband Code Division Multiple Access), to reach higher data rates in third generation network called UMTS. The access network in UMTS uses both circuit switched and packet switched connection. However, these connections are separated that circuit switched connection for real-time services and a packet switched connection for data communication services. The point take into the consideration is in UMTS, the IP address allocation of the user equipment (UE) is carried out when a data communication service is established [1]. Similarly, when the service is released, the IP address is released then. It shows that data communication services still tied to the circuit switched core for paging. Disperately, EPC is purely formed of Internet Protocol (IP) structure that both real-time and data communication services are carried by IP. Moreover, the IP address allocation of the user equipment has been done when the mobile is switched on and released when switched off.

The new access solution, LTE is based on OFDMA (Orthogonal Frequency Division Multiple Access) that provides orthogonality between the users, reducing the interference and improving the network capacity. Furthermore, the advantages are a combination with higher order modulation (up to 64QAM), large bandwidths (up to 20 MHz) and spatial multiplexing in the downlink (up to 4x4) that yields higher data rate. The highest theoretical peak data rate on the transport channel is 75 Mbps in the uplink, and in the downlink, using spatial multiplexing, the rate can be as high as 300 Mbps [14].

Figure 2.2 illustrates the EPC base stations evolved NodeB and interface relations internal and external gates.

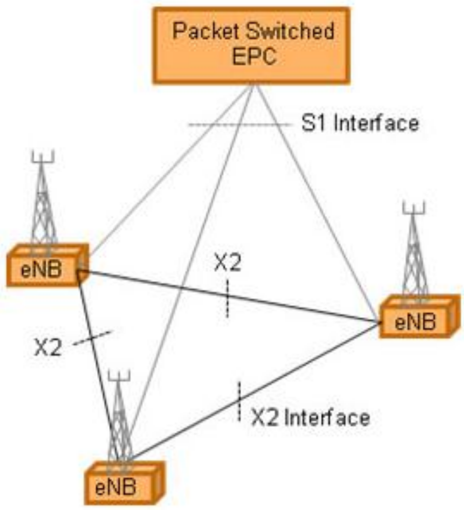

Figure 2.2 : S1 and S2 interfaces of LTE[36].

The eNodeBs are normally inter-connected via the X2-interface. Externally, these base stations are connected to the core network by the S1-interface. The remarkable difference is that there is no centralized intelligent controller like GSM and UMTS technologies. Since the controller intelligence are distributed among the base stations, the connection set-up and elapsed time for handover is reduced. For an end-user the connection set-up time for a real-time data session is in many cases crucial, especially in on-line gaming. The time for a handover is essential for real-time services where end-users tend to end calls if the handover takes too long.

2.2 Technologies Behind LTE Radio Access

Figure 2.3 indicates the LTE radio access overview from Release 8 to Release 10 that initial work in LTE technology relies on the late 2004 to provide new radio access technology focusing on packet switched data only. The first phase of the 3GPP work on LTE was to define a set of performance and capability targets for LTE. The targets are defined to maintain peak data rates, higher user and system throughput, spectral efficiency, and lower latency [14]. The most important technologies bring by LTE Release 8 are transmission schemes, scheduling, multi-antenna support and spectrum flexibility. Then, Release 8 is followed by additional LTE releases as LTE releases 9 and 10, introducing additional functionality and capabilities in different areas like positioning, relaying, multi antenna extension as illustrated in Figure 2.3.

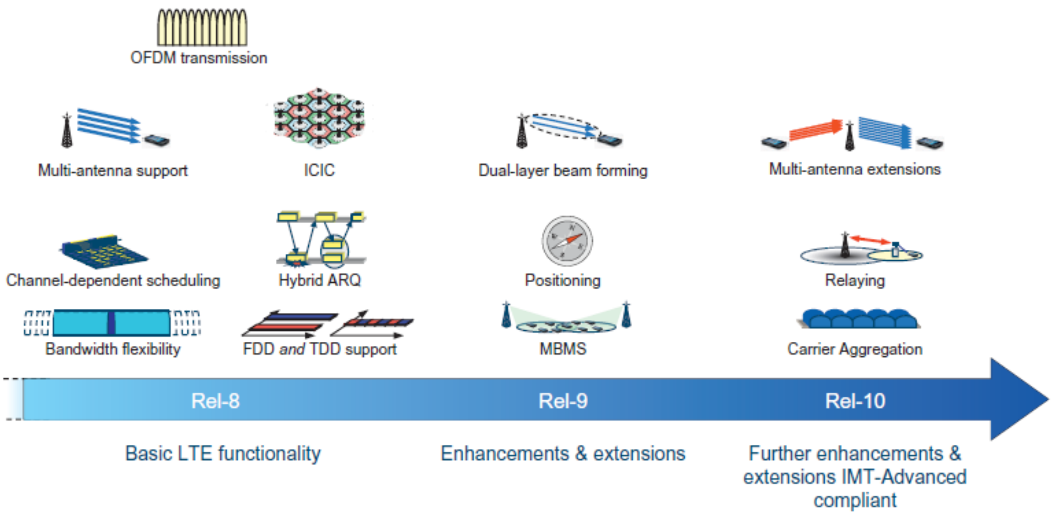

Figure 2.3 : LTE Radio Access Overview[14].

2.2.1 Orthogonal frequency devision multiple access (OFDMA)

The LTE downlink transmission scheme is based on OFDMA which is multi-carrier, multiple-access transmission technology [15]. Radio spectrum access is based on the OFDM scheme. In this technology, whole data is divided into smaller chunks, and each chunk of data is transmitted in parallel in many narrowband orthogonal subcarriers. OFDM is a special case of Frequency Division Multiplexing (FDM). In FDM whole data is sent in a single wideband carrier. Orthogonality of sub-carriers is very important in this transmission technique. If the subcarriers are not orthogonal to each other, it causes inter-carrier interference. The orthogonal subcarriers are shown in Figure 2.4.

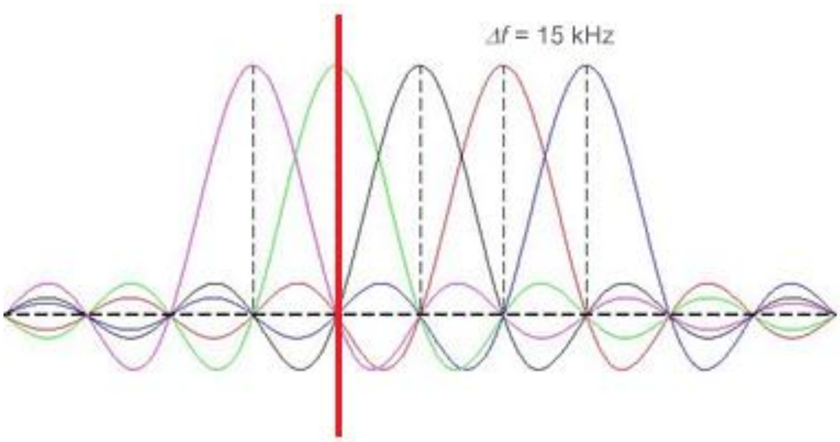

Figure 2.4 : OFDM Subcarriers[37].

Each subcarrier can be modulated with different modulation techniques in parallel, depending upon the channel condition of the particular subcarrier. The sub-carrier having good channel condition can go for higher order modulation like 64-Quadrature Amplitude Modulation (64-QAM), while the sub-carrier having bad channel condition has to adapt to QAM.

Differently from basic OFDM, OFDMA allow multiple access by assigning sets of sub-carriers to each individual user. It can exploit sub-carriers distributed inside the entire spectrum [16]. It means sub-carries are used for several users instead of a single user like in OFDM. In the case of OFDMA, subcarriers are multiplexed or combined into a larger set of subcarriers to which the IFFT operation is applied [17]. Information on which subcarriers belong to which user is also sent in the downlink.

The main advantage of OFDM modulation is its good performance in frequency selective channels. If a channel has deep fades at some frequencies, only the corresponding sub-carriers are affected. It also makes channel equalization easy in receiver side, where frequency domain equalization can be implemented. Furthermore, insertion of cyclic prefix makes it robust against the delay spreading of signal [17].

OFDM also provides some additional benefits relevant for LTE:

- OFDM is key enabler to provide operation in the frequency domain that allows channel-dependent scheduler to work compared to time domain only scheduling used in major 3G systems.
- From a baseband perspective, varying number of OFDM subcarriers used for transmission. It means, transmission bandwidth is flexible enough to support operation in spectrum allocations of different size.
- Moreover, for broadcast or multicast transmission OFDM provides to have the same information is transmitted from multiple base stations.

2.2.2 Channel dependent scheduling

Usage of OFDM in the physical layer of LTE provides that the scheduler can operate with both the time and frequency domains. It means, the scheduler can select users with the best channel conditions for each time instant and frequency region. The advantage is scheduler can schedule a terminal taking consideration of the channel in time and frequency domain. This is one of the main features of LTE. Figure 2.5 demonstrates the downlink scheduling in time and frequency domains.

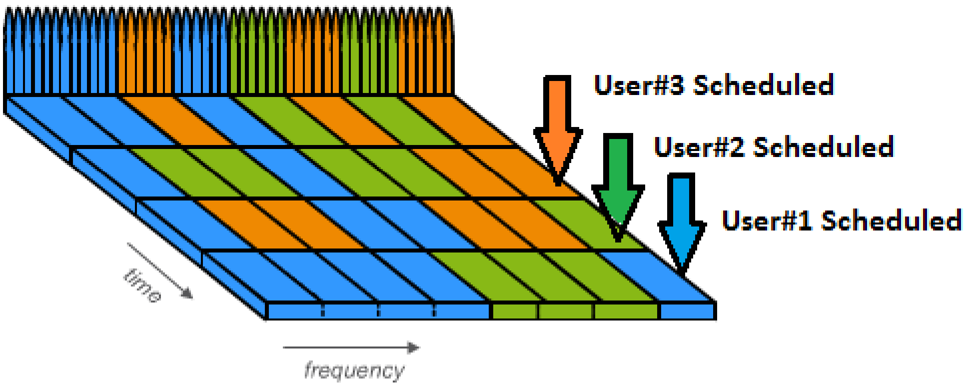

Figure 2.5 : Downlink Scheduling in time and frequency domains[37].

Channel-dependent scheduling in a mobile communication system deals with the question of how to share, between different users (different terminals), the radio resources available in the system to achieve as efficient resource utilization as possible [19]. Typically, this implies minimizing the amount of resources needed per user and hence allowing for as many users as possible in the system, while still satisfying their QoS requirements. According to the channel conditions, the scheduler can determine how many resource blocks should be allocated to a user, and what time slot and frequency band should be used. Furthermore, as to the channel condition the scheduler can also decide what data rate the terminal should use to transfer data. Better channel conditions enables higher data rates to be used.

Channel-dependent scheduling depends on channel quality variations between users to obtain a gain in system capacity. The frequency domain scheduling is suitable when the channel is varying slowly in time. However, for delay sensitive services time domain scheduler helps to schedule a particular user. For LTE, scheduling decisions can be taken as often as once every 1 ms and the granularity in the frequency domain is 180 kHz. This allows for relatively rapid channel variations in both the time and frequency domains to be followed and utilized by the scheduler. To support downlink scheduling, user equipments provide the network channel quality reports instantaneously in both time and frequency domains. The channel state reports are obtained by measuring on reference signals transmitted in the downlink. According to these reports, the downlink scheduler can assign resources for downlink transmission to different users considering the channel quality in scheduling decision. Principly, a scheduled user can be assigned an arbitrary combination of 180 kHz wide resource blocks in each 1 ms scheduling interval [14].

2.2.3 Bandwidth and spectrum flexibility

Bandwith flexibility is an important characteristic of LTE that provides different transmisson bandwidths in downlink and uplink. It means LTE deployments can vary between different frequency bands that allows operating in different spectrums which are paired and unpaired spectrum by supporting both Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD) as shown in Figure 2.6.

Usage of OFDM brings the possibility of frequency dependent channel scheduling and dynamic allocation of transmission bandwidth. Allocation of bandwidth is done in terms of resource blocks (RB) [19].

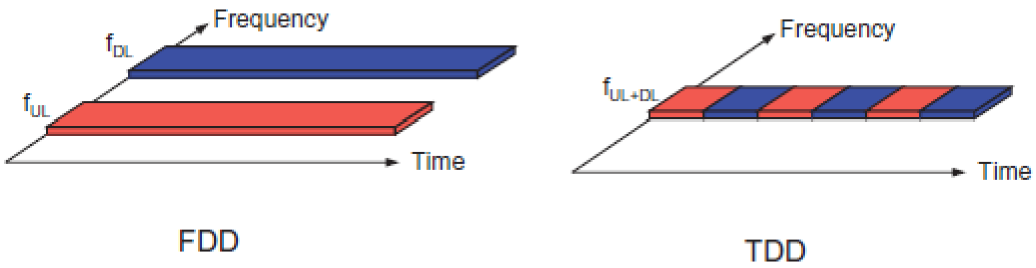

Figure 2.6 : Frequency and Time Devision Duplex[14].

The Table 2.1 indicates the available LTE channel bandwidth ranging from 1.4 MHz up to around 20 MHz and corresponding number of resource blocks.

Table 2.1 : Relation between channel bandwidth, corresponding resource blocks.

Channel Bandwidth (MHz)	Number of Resource Blocks
1.4	6
3	15
5	25
10	50
15	75
20	100

2.2.4 CQI reporting and AMC

The procedure of the Channel Quality Indicator (CQI) reporting is a fundamental feature of LTE networks [17] since it enables the estimation of the quality of the downlink channel at the eNodeB. Each CQI is calculated as a quantized and scaled measure of the experienced Signal to Interference plus Noise Ratio (SINR). The main issue related to CQI reporting methods is to find a good tradeoff between a precise channel quality estimation and a reduced signaling overhead.

The CQI reporting procedure is strictly related to the Adaptive Modulation and Coding (AMC) module, which selects the proper Modulation and Coding Scheme (MCS) trying to maximize the supported throughput with a given target Block Error Rate (BLER) [17]. In this way, a user experiencing a higher SINR will be served with higher bitrates, whereas a cell edge user or in general a user experiencing bad channel conditions will maintain active connections, but at the cost of a lower throughput.

In section 2.3.4 Resource Allocation part, more explanation is held about the CQI reporting and AMC related with the state of work.

2.2.5 Hybrid ARQ and Soft-Combining

Hybrid ARQ (HARQ) is combination of Automatic Repeat- reQuest (ARQ) and Forward Error Correction (FEC). ARQ is used for retransmission purposes, in which receiver sends Acknowledged (ACK) or Not Acknowledged (NACK) corresponding to the correct or incorrect message it receives [18]. The receiver uses Cyclic Redundancy Check (CRC) to check the accountability of the received message. The terminal can correct some erroneous messages itself. For the correction, it uses some Forward Error Correction (FEC) codes such as Turbo codes. Moreover, the receiver combines the previously received erroneous messages while retransmitting the new messages. The process is called soft combining. With soft combining, the energy per bit of the received message can be increased [18]. Retransmission is held at two levels in LTE that are Radio Link Control (RLC) level retransmission and Medium Access Control (MAC) level retransmission. Only MAC layer supports HARQ; the RLC layer supports ARQ only.

2.3 LTE System Architecture

3GPP standardize LTE with two part formed with radio-access and core network which are called LTE RAN (LTE- Radio Access Network) and EPC (Evolved Packet Core) respectively [36] as shown in Figure 2.7. The LTE RAN is responsible for all radio related functionality of the overall network including, for example, scheduling, radio-resource handling, retransmission protocols, coding and various multiantenna schemes. Besides, the EPC is responsible for functions not related to the radio interface but needed for providing a complete mobile-broadband network. This includes, for example, authentication, charging functionality, and setup of end to end connections. Handling these functions separately, instead of integrating them into the RAN, is beneficial as it allows for several radio-access technologies to be served by the same core network.

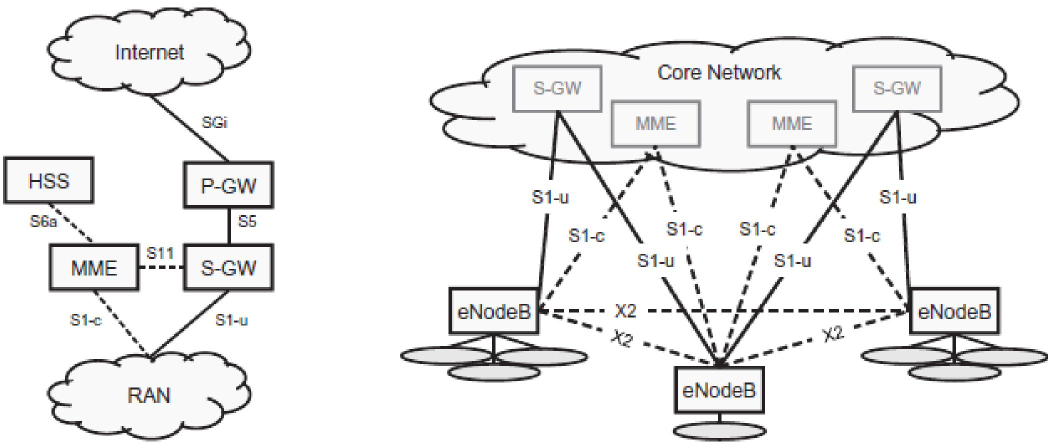

Figure 2.7 : LTE RAN - EPC Architecture[14].

However since this thesis related with the QoS aware scheduling based on radio access side of the LTE network, primarily radio access downlink protocols, traffic flows, scheduling models are explained followingly.

2.3.1 Radio access protocols

Figure 2.8 shows the radio protocol architecture for LTE downlink case. For IP packets provided by RAN, PDCP (Packet Data Conversion Protocol) adds some header to the packets for in-sequence delivery, and also applies some header compression algorithm to reduce overheads [16]. PDCP also does encription of the data. RLC (Radio Link Control) does segmentation of the packets passed by PDCP to it. Its responsibility also includes simple retransmission handling (it does not support HARQ) and in-sequence delivery of the packets. The RLC provides services to the PDCP in the form of radio bearers. HARQ resides in MAC layer. Scheduling is handled in this layer. The MAC provides services to the RLC in the form of logical channels.

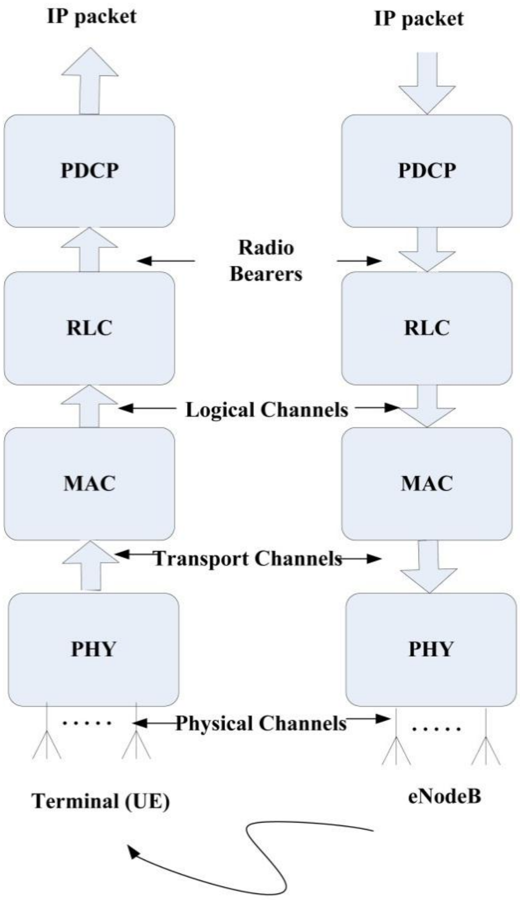

Figure 2.8 : LTE Downlink Layer Flow.

PHY (Physical Layer) is the lowest layer in the protocol architecture. Its main responsibility is handling coding, modulation, and antenna mapping. The PHY provides services to the MAC in the form of transport channels. Different layers at the receiver side reverse process the works done by corresponding layers at the transmitter. For example, functions of PHY layer at receiver side are de-mapping, demodulation, and decoding to lists different kinds of channels at different protocol layers of LTE, both in uplink and downlink directions. Furhermore, Table 2.2 includes the related channels for logical, transport and physical channels in both directions. However, this thesis mainly relies on MAC layer.

Table 2.2 : LTE Channel Types.

	LTE Downlink Channels	LTE Uplink Channels
Logical Channels	PCCH, BCCH, CCCH, DTCH, DCCH, MTCH, MCCH	CCCH, DTCH, DCCH
Transport Channels	PCH, BCH, DL-SCH, MCH	UL-SCH, RACH
Physical Channels	PBCH, PDSCH, PDCCH, PHICH, PCFICH, PMCH	PUSCH, PUCCH, PRACH

2.3.2 MAC layer

In this section MAC layer and its state of work for scheduling is explained. MAC layer primarily works on multiplexing of logical channels, HARQ retransmissions, and uplink and downlink scheduling [18]. The scheduling functionality of the MAC takes an important role for this thesis that is located in the eNodeB for both uplink and downlink channel. Moreover, the HARQ protocol part of MAC layer presents the transmitting and receiving ends of the MAC protocol. After the processing of data in MAC layer, MAC provides services to the RLC in the form of logical channels.

Inside the eNodeB, MAC is responsible for multiplexing of different logical channels and mapping them to the appropriate transport channels. In addition to multiplexing of different logical channels, the MAC layer can also insert the MAC control elements into the transport blocks to be transmitted over the transport channels. The MAC control element is used for inband control signaling for example, timing-advance commands and random-access response. As can be seen in the Figure 2.9, these control elements are identified with reserved values in the Logical Channel Index (LCID) field, where the LCID value indicates the type of control information. The LCID field in one of the MAC subheaders is set to a reserved value indicating the presence of a buffer status report [14]. From a scheduling perspective, buffer information for each logical channel is beneficial. The buffer size field in a buffer-status report indicates the amount of data awaiting transmission across all logical channels in a logical channel group. The buffer-status report can be triggered periodically when the arrival of data has higher priority that the one that is currently in the transmission buffer. This process affects the scheduling decision. Secondly, if the user begin to be served by another cell, buffer-status report will provide about the new cell information and situation of the user.

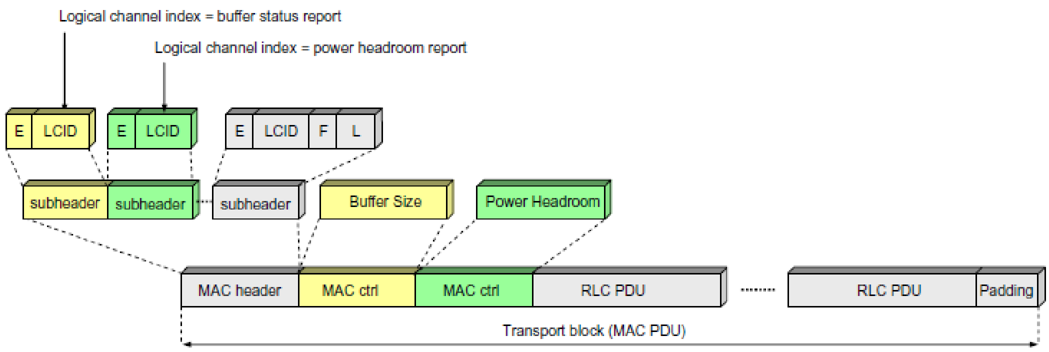

Figure 2.9 : Transport Block Status Reports[14].

From the physical layer, the MAC layer uses services in the form of transport channels [13]. A transport channel shows the information characterictics during transmission over the radio interface. This information on a transport channel is organized into transport blocks. In each Transmission Time Interval (TTI), at most one transport block is transmitted over the radio interface from a terminal in the absence of spatial multiplexing. In the case of spatial multiplexing (MIMO), there can be up to two transport blocks per TTI. The transport format which is defined for each transport block includes information about the transport-block size, the modulation and coding scheme, and the antenna mapping. By varying the transport format, the MAC layer can thus realize different data rates. Rate control is therefore also known as transport format selection. The MAC layer in data link layer of Open System Interconnection (OSI) aims to control the authority of user about the accessing media and resource [14].

Scheduler is an important issue in MAC layer for the system performance could be improved by an efficient scheduling algorithm to assign the priority of each user. For this reason, scheduler in MAC layer is the main factor to effects the system performance and resource reusability.

2.3.3 Downlink frame structure

In LTE transmission, physical radio resources are allocated into the time and frequency domain. In frequency domain, LTE network exploits OFDMA at the downlink that provides using narrow-band subcarriers distributed in the entire spectrum. LTE is a flexible radio interface that using varying bandwidths ranging from 1.4 MHz to 20 MHz that brings higher data rates [13]. In time domain, time is divided into frames and sub-frames that each subframe is 1ms. Then, each subframe is further divided into two slots each 0.5 ms, making the total time for one frame equivalent to 10 ms. Each time slot on the LTE downlink system consists of 7 OFDM symbols. The physical resources of the LTE downlink can be illustrated using a frequency-time resource grid as shown in Figure 2.10.

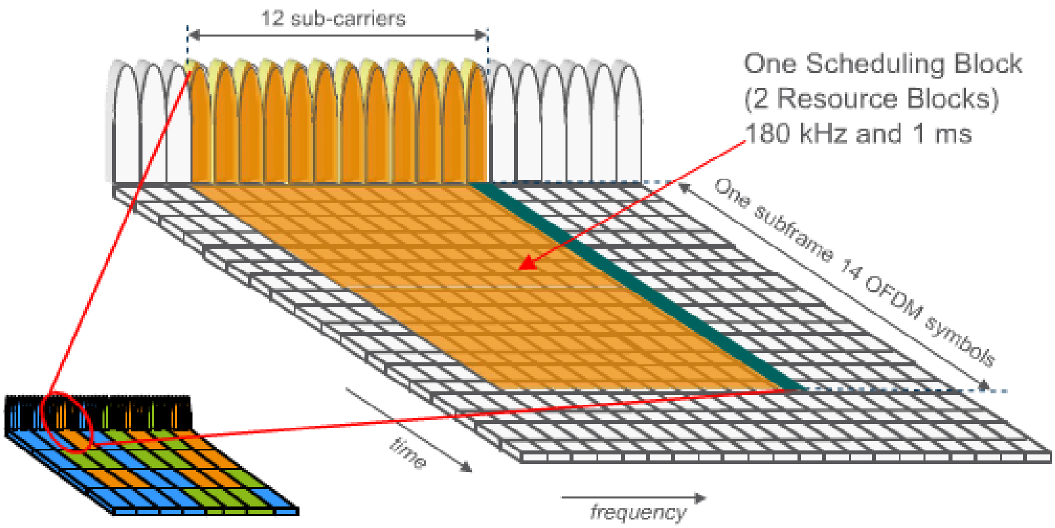

Figure 2.10 : LTE Downlink Frame Structure[38].

- 1 Resource Element (RE) = 1 subcarrier during 1 OFDM Symbol
- 12 OFDM symbol x 15 kHz = 180 kHz
- 180 kHz x 0.5 ms = 1 RB (resource block)
- 180 kHz x 1 ms = 2 RB (scheduling block)

In Figure 2.11, slot frame structure of the LTE network is illustrated.

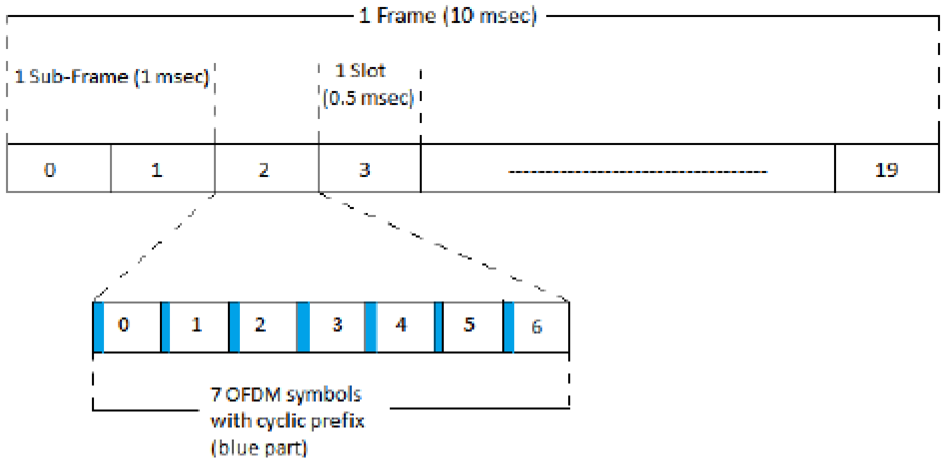

Figure 2.11 : LTE Frame - Slot Structure[14].

In LTE network, a resource block (RB), which is the minimum resource unit assigned to User Equipment (UE), corresponds to the bandwidth of 180kHz in frequency domain and 0.5ms in time domain [13]. In this context, scheduler becomes the key element which located at the base station and it assigns the resource block to the terminals.The scheduler determines for each 1 ms sub-frame which user(s) are allowed to transmit and on which resources are allocated.

In order to maintain orthogonality of the subcarriers, the guard period is generated as a cyclic prefix (CP), by repeating some samples from the end of each symbol at the beginning [15] as shown in Figure 2.12.

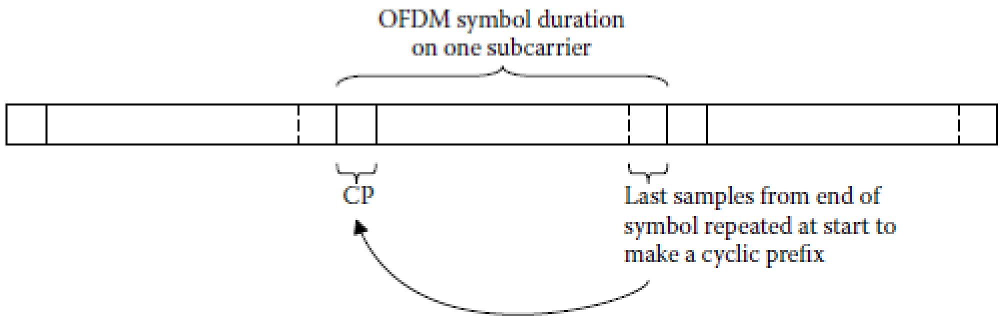

Figure 2.12 : Guard Period Cycle Prefix.

According to chosen bandwidth from 1.4Mhz to 20MHz, number of available RB is varied from 6 to 100 respectively. It is straightforward to see that each RB has 84 resource elements in the case of normal cyclic prefix and 72 resource elements in the case of extended cyclic prefix [13].

2.3.4 Resource allocation

In the scheduling process, the selection of the Modulation and Coding Schemes (MCS) is a key issue. In LTE downlink Adaptive Modulation and Coding (AMC) technique is used to select an appropriate MCS level for UE's transmission according to user channel state or condition [19].

Firstly, the UE estimates the instantaneous channel condition to determine Channel Quality Indicator (CQI) index. Then, UE sends this CQI report as a feedback to the eNodeB. Consequently, the eNodeB uses this CQI index to choose a certain level of MCS and allocates resource block pairs accordingly to the UE. The relationship between CQI index and the MCS is shown in Table 2.3, where larger CQI indices represent higher MCS levels. Moreover, the table demonstrates more efficient usage of channel resources and fewer resource block pairs occupied for transmission.

Table 2.3 : Mapping of CQI values versus MCS.

CQI Index	Modulation	Code Rate x 1024	Efficiency	CQI Index	Modulation	Code Rate x 1024	Efficiency
0	Out of range			8	16QAM	490	1.914
1	QPSK	78	0.152	9	16QAM	616	2.406
2	QPSK	120	0.234	10	16QAM	466	2.730
3	QPSK	193	0.377	11	64QAM	567	3.322
4	QPSK	308	0.602	12	64QAM	666	3.902
5	QPSK	449	0.877	13	64QAM	772	4.523
6	QPSK	602	1.176	14	64QAM	873	5.115
7	16QAM	378	1.477	15	64QAM	948	5.554

However, differences in modulation and coding rates produces different Block Error rate (BLER) performances. The relationship between Signal to Noise Ration (SNR) and BLER curves of each CQI index were studied extensively in the literature [19] indicating clearly that CQI is a reflection of channel quality as SNR, where a high CQI reflects good channel condition, under which a high level of MCS could be selected to achieve high bit rate and efficiency [20].

Figure 2.13 represents the main resource allocation modules that interact with the downlink packet scheduler.

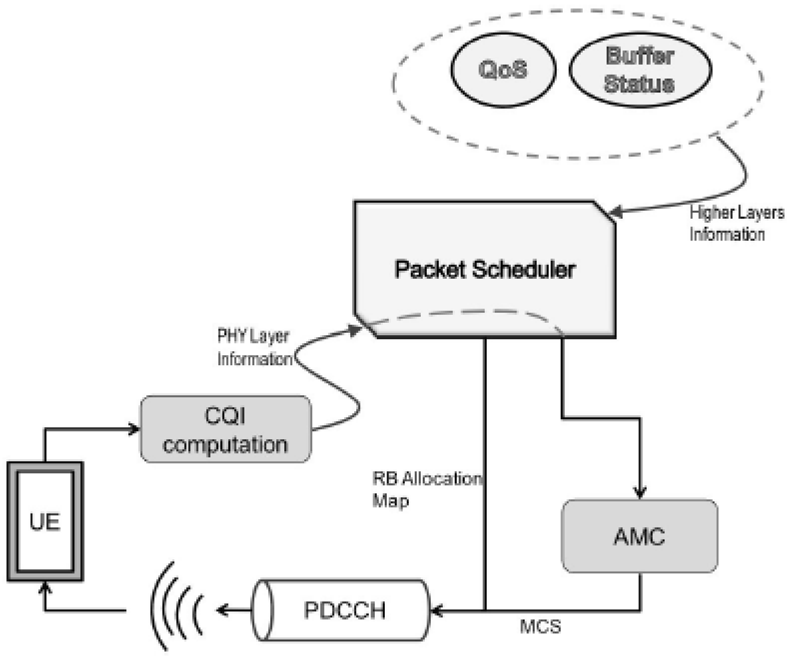

Figure 2.13 : Simplified Model of Resource Allocation[21].

The entire process shown in Figure 2.13 is repeated in every every TTI. The initial step each UE decodes the reference signals, computes the CQI. Therby, UE sends this CQI information as feedback to the eNodeB. Secondly, the eNodeB uses the CQI information for the allocation decisions. Thereafter, the AMC module selects the best MCS. These information about these users which are the allocation of RBs and the selection of MCS are sent to the UEs on the Physical Downlink Control Channel (PDCCH). Then each UE reads the PDCCH payload and, in case it has been scheduled, accesses to the proper Physical Downlink Shared Channel (PDSCH) payload [21].

3. SCHEDULING ALGORITHMS

In LTE networks, a dynamic scheduling is a key process to provide efficient usage of limited resources with the lowest cost. A resource block (RB), which is the minimum unit assigned to terminals, corresponds to the bandwidth of 180kHz in frequency domain and 0.5ms in time domain. For the downlink direction, if the bandwidth is 20 MHz, there are 100 RBs available in the system. When there are 50 active users, there are 100^{50} different RB to UE assignments. This huge possibility shows the scheduler importance and ability. Especially for highly loaded networks, scheduler can significantly affect the system performance.

We also know that the key characteristic of the mobile radio communication is rapid and significant changes in the channel conditions due to shadowing, fast fading, multipath, etc. These instantaneous and random variations affect the quality of each radio channel in a cell. Therefore, the changes in channels should be taken into account when scheduling is performed. Thus, our objective in this thesis is to propose a novel scheduler to maximize the network performance.

Scheduling controls the allocation of shared resources among the users at each time constant. In LTE, for each Transmission Time Interval (TTI) of 1 ms, the scheduler determines which user(s) are allowed to transmit and which resources are allocated. After scheduling, scheduling map will be sent to all terminals. Individual terminal only decodes received data in certain particular time and frequency based on scheduling map. By periodically receiving the feedback information from the terminals, the scheduler in the base station dynamically adjusts scheduling decisions to ensure efficient resource allocation and quality of service requirements.

Generally, scheduling can be divided into two classes: channel-independent scheduling and channel-dependent scheduling [26]. Channel-independent scheduling does not take channel conditions into account. On the contrary, channel-dependent scheduling is performed by allocating resources based on channel conditions. It deals with how to share available radio resources between different terminals to achieve efficient resource utilization as possible. This implies to minimize the amount of resources needed per user while still satisfying quality of service requirements. Thus, as many users as possible in the system can use the resources.

Furthermore, scheduling can be classified based on application scenarios, since the performance of scheduling algorithms highly relies on the type of incoming flows. In order to achieve application specific performance, it is important to select suitable algorithms according to traffic types of flows which can be divided into real-time and non real-time by considering LTE services, including such as voice, data and video [28].

In this chapter, existing scheduling algorithms considering traffic types and channel qualities are explained below along with their main ideas. Followingly, Chapter 4 represents the proposed algorithms which are focus on channel-dependent scheduling to achieve better performance in resource allocation between real-time and nonreal-time users.

3.1 Non Real Time Scheduling Algorithms

A packet scheduler performs its allocation decision to maximize the satisfaction level of system requirements. A scheduler measures system satisfaction based on a desirable performance metric, such as per UE's experienced data rate, fairness in resource allocation among UEs, average packet delay experienced by UEs, etc. The choice of what performance metric to optimize influences how the scheduler resolves resource contention among UEs. In general, the scheduling algorithms aim at providing efficient resource sharing, better performance in terms of delay, throughput, link utilization, fairness, complexity, etc. Scheduler pursues the optimal or suboptimal solution of above functions, while considers computation overhead.

The most popular scheduling algorithms for nonreal-time flows in LTE networks are Round Robin [25], MaxC/I [26] and Proportional Fair [29]. The performance of nonreal-time algorithms is measured with throughput and fairness.

In the following subsection we will provide more detailed discussion on studied three scheduler algorithms.

3.1.1 Round robin

Round Robin is one of the simplest resources scheduling algorithms. It is commonly applied in operating systems and computer networks. In the beginning, terminals are ordered randomly in a queue. The new terminals are inserted at the end of the queue. The first terminal of this queue is assigned all the available resources by scheduler, and then put it at the rear of the queue. The rest of step sound follow the same way, until no terminal applies for resources. On one hand, it is a fair scheduling, since every terminal is given the same amount of resources. On the other hand, it neglects the fact that terminals in bad channel conditions need more resources to carry out the same rate, so it is not absolutely fair. However, it is much fairer than Max C/I. Although Round Robin gives every terminal equal chance to obtain resources, the overall throughput is much lower than Max C/I, because scheduler does not consider channel conditions.

Robin scheduling is a non-aware scheduling scheme that lets users take turns in using the shared resources (time/RBs), without taking the instantaneous channel conditions into account [29]. Therefore, it offers great fairness among the users in radio resource assignment, but degrades the system throughput performance.

In LTE, different terminals have different service with different QoS requirements. It is impossible to allow every terminal to take up the same resources in the same possibility, because it will decrease efficiency of resources. Nevertheless, Round Robin is a good reference to measure the fairness of scheduling for LTE.

Fig 3.1 illustrates the Round Robin, MaxC/I and Proportional Fair algorithms' resource allocation as a ring chart flow.

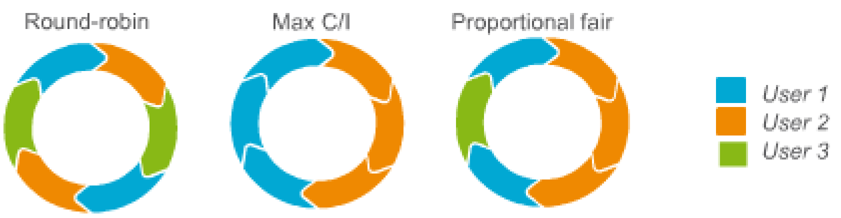

Figure 3.1 : RR, Max C/I, PF Schedulers[37].

3.1.2 Max C/I

As the name implies, this scheduling strategy assigns RBs to the UE with the best radio link conditions. In order to perform scheduling, the channel quality indicator (CQI) reports are generated by the UE and fed back to the eNodeB in quantized form periodically. Scheduler analyzes CQI values from terminals to obtain data rate of an identical sub-channel for different terminals. Then scheduler assigns this sub-channel to the terminal which can achieve the highest data rate in this sub-channel based on SNR. A higher CQI value means better channel condition. Based on the CQI received, the best CQI is selected for scheduling. The following equation describes the Max C/I that $R_{k,n}$ (t) represens data rate of terminal k for one sub-channel n in time slot t **(3.1)**.

$$i = \arg\max_{k} R_{k,n}(t) \tag{3.1}$$

As a channel dependant scheduling, Max C/I takes advantage of multiuser diversity to carry out maximum system throughput[28]. However, terminals in bad channel conditions are never considered by this scheduler algorithm that causes starving terminals while having best channel terminals obtain maximum throughput. Therefore, this scheduler is not a fair algorithm. Nevertheless, it can be considered as a good reference to distribute resources among terminals while they realy acquire maxcimum throughput rates. In this thesis, Max C/I is used as mainly reference scheduler algorithm to allocate resource blocks to the users.

3.1.3 Proportional fair

Proportional Fair is a compromise between Maximum Rate and Round Robin [29]. It pursues the maximum rate, and meanwhile assure that none of terminals is starving. The terminals are ranked according to the priority function. Then scheduler assigns resources to terminal with highest priority. These steps are executed recursively until all the resources are used up or all the resources requirements of terminals are satisfied.

The following equation describes the priority function that $R_{k,n}$ (m) represens the estimation of the supported data rate of terminal k for resource block n where R_k (m) shows the average data rate of terminal k over a windows in the past **(3.2)**.

$$i = \arg\max_k \frac{R_{k,n}(m)}{R_k(m)} \quad \textbf{(3.2)}$$

The update of the R_k in this scheduler can be specified in two ways. One of them is that R_k is updated when all resource blocks are allocated. The other one is that R_k is updated after each resource block allocation. **(3.3)** is the equation to present R_k update where T shows the windows size of the average throughput if the user k is selected. In our simulation environment, PF scheduler takes the second update prosedure during allocation.

$$R_k(m+1) = (1-\frac{1}{T})R_k(m) + \frac{1}{T}R_{k,n}(m) \quad \textbf{(3.3)}$$

The PF provides a good trade off between system utility and fairness by selecting the user with highest instantaneous data rate relative to its average data rate [28], [29]. The PF was designed specifically for the BE service class and hence does not guarantee any QoS requirement such as delay, jitter and latency.

In the literature, many approaches are proposed to define fairness parameter; one of the most outstanding parameter is Ray Jain's fairness index [27]. Thus, in this thesis Jain's fairness index, **(3.4)** is used to measure fairness among users in the system.

$$J(\overline{T}) = \frac{(\sum_{k=1}^{K} T(k))^2}{K\sum_{k=1}^{K} T(k)^2} \quad \textbf{(3.4)}$$

It is a throughput and number of UEs based equation where K is the total number of users and $\bar{T}$ is a vector representation of user equipments' current throughput values.

The fairness index changes between zero and one. When whole users have the same throughput in the system, the fairness index becomes one that it is the ideal fairness scenario.

3.2 Real Time Scheduling Algorithms

Real-time flows have more delay sensitive than nonreal-time flows. Therefore the scheduling algorithms are differentiated resulting in the reduction of influence of error correction.

In wireless communication, RT flows can be modeled as arrival process of independent packets to respective queues, under a delay target constraint. For different wireless network models, there are several stabilizing policies to satisfy different delay target constraints [28]. For example, the stabilizing policy of system whose delay target is the maximum delay deadline is to make sure that the length of queues is in a certain bound.

The most popular scheduling algorithms for RT flows are Largest Delay First, Modified Largest Weighted Delay First. The main performance metrics are firstly delay experience then packet loss rate and fairness for RT purpose.

3.2.1 Largest delay first

This scheduler is very similar to the Round Robin algorithm for NRT flows. It neglects the channel condition and when the first package is received by base station, it will be sent out in the first place. According to equation **(3.5)**, D_k is the time that package of terminal k has spent at the base station waiting for scheduling.

$$i = \arg\max_{k} D_k(m) \tag{3.5}$$

However, because of this scheduler first in first out characteristic property, result becomes with poor throughput.

3.2.2 Modified largest weighted delay first

Modified Largest Weighted Delay First Algorithm is designed to balance the delay of the packects to use the channel resources efficiently. The M-LWDF resource allocation algorithm is tailored to support multimedia services, considering that this scheduler outperforms other state-of-the-art schedulers in supporting video streaming services [30].

This scheduler mainly relies on the delay of the packets: the packets that violate the target delay of the service while waiting at the MAC buffer will be discarded. However, NRT services are scheduled using the classical PF rule. The MLWDF was designed to meet QoS requirements and has been proved to be throughput-optimal in [31]. A scheduling rule is termed throughput optimal if it satisfies the rule that it renders the queues to be stable.

(3.6) describes the priority function of the MLWDF scheduler where R_k (m) shows the average data rate of terminal k, D_k is the delay of the first packet to be transmitted in the queue and α_k is the maximum probability that the packet delay exceeds the target delay.

$$i = \arg\max_{k} \alpha_k * D_k(m) * \frac{R_{k,n}(m)}{R_k(m)} \qquad \textbf{(3.6)}$$

Although this thesis has not had enough resource to extend our algorithms to compare these algorithm results, MLWDF are studied here to prepare for the future work in Section 6.

3.3 Mixture of Real Time and Non Real Time Scheduling Algorithms

In realistic networks RT and NRT traffics are flowing together and both of them are considered with mixture real-time and nonreal-time scheduling algorithms. The most popular scheduling algorithms for mixture of real-time and nonreal-time flows are Exponential Rule, Virtual Token Modified Largest Weighted Delay First. The QoS performance parameters are measured by throughput, delay experience and fairness.

3.3.1 Exponential proportional fair

It is important to ensure that a proper trade-off is maintained between fairness, throughput-optimality and delay requirements for a scheduling algorithm to be truly efficient. The Exponential Rule algorithm is designed to achieve this goal. The intuition is to increase the weight of the far away users on slight improvement in their channel conditions while keeping the weight of the users near by proportionately less since they would get very good data rates due to their proximityis designed to balance the delay of the packects to use the channel resources efficiently [30]. The exponential rule is given by **(3.7), (3.8), (3.9)**:

$$i = \arg\max_{k} \gamma_k R_k(t) \exp\left(\frac{\alpha_k W_k(t) - \overline{\alpha W}}{1 + \sqrt{\overline{\alpha W}}} \right) \tag{3.7}$$

$$\overline{\alpha W} = \frac{1}{N} \sum_k \alpha_k W_k(t) \tag{3.8}$$

$$\gamma_k = \frac{\alpha_k}{\overline{R_k(t)}} \tag{3.9}$$

R_k (t) shows estimation of supported data rate of terminal k in time slot t. $\overline{R_k(t)}$ is the average data rate of terminal k. α_k is the priority coefficient which reflects the desired QoS of terminal k. W(t) is the longest time that terminal k's packets are spent on base station. When the differences in the waiting times are very small, a small change in the waiting time is reflected as a large change in the exponent value due to which, the rule behaves like the PF algorithm.

3.3.2 Virtual token modified largest weighted delay first

Virtual Token Modified Largest Weighted Delay First Algorithm is designed to balance the delay of the packects to use the channel resources efficiently. The virtual token resource allocation algorithm,VT-M-LWDF, focuses on RT services whereas a minimum throughput has been guaranteed for NRT services [31].

This scheduler is mainly based on the queue size of each RT flow‘s buffer, as this helps allocate most of the resources to video flows first, whereas NRT services are penalized. The virtual token scheme is a modification of the M-LWDF rule that VT-M-LWDF scheduler takes into consideration, in addition to the earlier parameters, the queue size parameter of each RT flow‘s buffer. This provides the scheduler an indication about the significance of the flow available in the buffer. Conversely, the M-LWDF scheduling decision is made with regard mainly to the packet delays.

The main goal of the VT-M-LWDF rule is to improve the QoS performance metrics for multimedia services, such as video and VoIP, and to maintain minimum throughput guarantee for NRT services. Furthermore, the latter impacts on the QoS requirements that belong to NRT services. Hence, there is a trade-off in the delivery of the QoS requirements between the RT and the NRT services. The following equation **(3.10)** illustrates the metric used to represent the VT-M-LWDF scheduler:

$$W_{k,n}(m) = \alpha_k * Q_k(m) * \frac{R_{k,n}(m)}{R_k(m)} \qquad \textbf{(3.10)}$$

$Q_k(m)$ refers to the queue size of the *k*-th flow at a particular scheduling epoch (m) when serving RT users. On the contrary, the PF scheduler is selected to make scheduling decisions when serving NRT users.

4. PROPOSED SCHEDULING ALGORITHMS

The task of a scheduler in an LTE base station is to allocate available physical resource blocks to the users. This allocation process is carried out within evolved NodeB (eNodeB) in the LTE network architecture. An eNodeB includes the functionalities of both Base Station Controller (BSC) and Radio Network Controller (RNC) which were two distinct entities in the traditional cellular network architecture. Radio resource allocation task, which distributes available radio resources to UEs, is carried out within eNodeB. None of the existing standard scheduling algorithms such as Round Robin, MaxC/I and Proportional Fair takes channel qualities and traffic types into account at the same time when the available resources are allocated to UEs. For example, MaxCQI scheduler allocates Resource Blocks (RBs) to UEs solely using Channel Quality Indicator (CQI) information without taking the QoS requirements of UEs into account. These schedulers also do not provide any instrument to influence the scheduling operation according to changing network conditions. Similarly, recent studies on the QoS-aware LTE schedulers can be found in Section 1.2 Literature Review part.

Self-Organizing Network (SON) is a type of network which optimizes the system parameters both at the deployment phase (planning stage) and also with respect to changing network conditions during run-time (e.g., changes in traffic demand, wireless link qualities, user locations, etc.). A SON feature for the LTE network can be realized in two steps:

(i) extending the existing LTE scheduling algorithms to make them reconfigurable to varying network conditions

(ii) reconfiguration of the LTE scheduling parameters when network conditions changes.

In this thesis, three LTE downlink scheduling algorithms, taking both channel qualities and traffic types of UEs into account, are proposed to realize the first step of the SON feature. A reconfigurable parameter in the proposed schedulers determines the resource prioritization level for real-time services, and hence allows network administrators to more effectively control the resource allocation according to the QoS requirements such as throughput, packet loss, delay and fairness of UEs and changing network dynamics.

Figure 4.1 demonstrates a high level design of our scheduling algorithm. First, incoming traffic is classified into real-time and nonreal-time flows. Active UEs, which need a resource block, are inserted into a priority list. The priorities are determined using traffic type and channel quality information of UEs. For example, real-time traffic has strict priority over nonreal-time traffic in one variant of the proposed scheduling algorithms. Finally, the scheduler assigns available resource blocks to the users according to the order in the priority list.

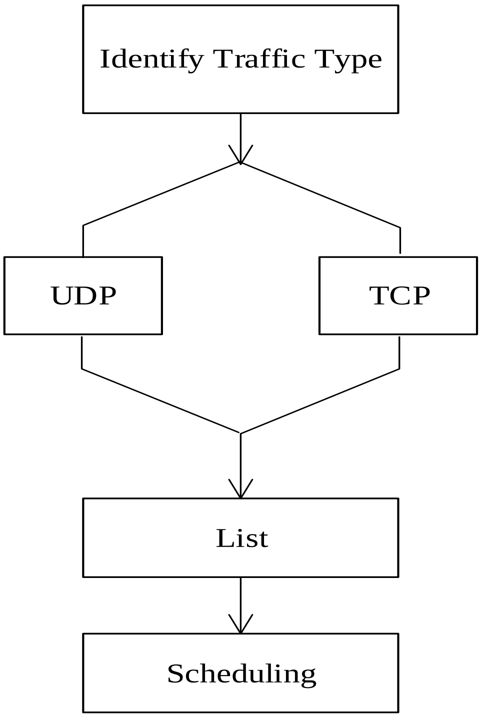

Figure 4.1 : Scheduling Algorithm Flow Chart.

The pseudo code of the first variant of our proposed scheduling algorithm (Algorithm_1) is shown below. The incoming traffic user is inserted to the priority list by using traffic type (real-time or nonreal-time) and channel quality information. In this version, all real-time flows have higher priority than nonreal-time flows and hence the traffic type is the most important factor to determine the insertion point of the incoming traffic in to the priority list.

Algorithm_1: Strict Priority based RB allocation for real-time traffic

```
N = Number of UEs;
CQI_i = Channel Quality Indicator for UE i;
T_i = Traffic Type for UE i;
RB = Available Resource Block;
L = Priority List for RB Allocation;
initialize L to empty
 for i=1 to N
     if (T_i == UDP)
         add UE i to upper part of L,
          insertion point to L is determined by CQI_i
    else if (T_i == TCP)
          add UE i to lower part of L,
          insertion point to L is determined by CQI_i
allocate RBs to UEs according to the order in L
```

It is conjectured that Algoritm_1 maximizes the performance of real-time traffic by providing higher throughput, lower loss rates, and lower end-to-end delays while the performance of nonreal-time traffic significantly degrades.

Algorithm_2: RB allocation with reconfigurable traffic prioritization

```
N = Number of UEs;
CQI_i = Channel Quality Indicator for UE i;
T_i = Traffic Type for UE i;
RB = Available Resource Block;
L = Priority List for RB Allocation;
X = Percentage of nonreal-time users whose priorities are made equal to real-time
users;
initialize L to empty
 for i=1 to N
       if (T_i == UDP)
          add UE i to upper part of L,
           insertion point to L is determined by CQI_i
       else if (T_i == TCP)
          add UE i to lower part of L,
          insertion point to L is determined by CQI_i
move X% of TCP flows to upper part of L,
          insertion point to L is determined by CQI values
allocate RBs to UEs according to the order in L
```

The pseudocode of the second variant of our proposed scheduler algorithm, Algorithm_2, is shown above. This is an improved version of Algorithm_1 in a sense that the network administrator can control the tradeoff between the performance improvement of UDP flows and the performance degradation of TCP flows by tuning the reconfigurable traffic prioritization parameter X. When X is set to zero, Algorithm_2 and Algorithm_1 are the same.

Algorithm_3 : RB allocation with reconfigurable traffic prioritization

---

N = Number of UEs;
CQI_i = Channel Quality Indicator for UE *i*;
T_i = Traffic Type for UE *i*;
RB = Available Resource Block;
L = Priority List for RB Allocation;
X = Percentage of nonreal-time users whose priorities are made equal to real-time users;
initialize *L* to empty
 for i=1 to *N*
        if (*T_i* == VoIP)
           add UE *i* to upper part of *L*,
             insertion point to *L* is determined by *CQI_i*
        if (*T_i* == Video)
           add UE *i* to lower part of *L after VoIP users*,
            insertion point to *L* is determined by *CQI_i*
       else if (*T_i* == TCP)
           add UE *i* to lower part of *L*,
            insertion point to *L* is determined by *CQI_i*
move X% of TCP flows to upper part of *L*,
            insertion point to *L* is determined by CQI values
allocate *RB*s to UEs according to the order in *L*

From the result of real-time and nonreal-time flow algorithms, Algorithm_3 is formed to serve multi traffic scenario including the administrative control aspect. This algorithm can also handle multi real-time traffics (VoIP and Video) with the reconfigurable performance parameter X.

The pseudocode of our proposed scheduler Algorithm_3 is shown above. Additionally, Algorithm_3 has advantages both supporting the multi-traffic scneraio and treating the real-time and nonreal-time traffic according to specified QoS parameters. With the help of this administrative operation, although the origin of the existing algorithms are only designed for nonreal-time traffics, Algorithm_3 can handle both real-time and nonreal-time traffic and even multi-traffic.

In this instance, proposed algorithm takes a crucial role to balance real-time traffic to gain better performance results internally while controlling the nonreal-time as well. In order to attain QoS for VoIP traffic, VoIP users are evaluated at the top of the priority list. Video streaming and the users has nonreal-time traffic, FTP, can be operated by the network administrater by using traffic type and channel quality information. It means, number of non real-time users with higher CQI values can be moved to planned area to be assessed in real-time traffic by prioritization parameter X. The tradeoff between the performance advantage of the UDP flows and the disadvantage of the TCP flows still exist, however the parameter X turns the undesired conditions to positive way if it is used in the right place in the right time. It denotes that X parameter is an adjustable parameter that can fullfill the expectation of network requirements.

Quality of Service (QoS) refers to the capability of a network to guarantee a certain level of performance to a selected traffic flow using various technologies [32]. In LTE networks, quality of service is measured quantitatively by related performance measures of each network service such as throughput, packet loss, delay, jitter, and fairness.

Throughput measurement in LTE downlink depends on the amount of information transferred from a server to a user equipment within a certain time. For example, for the file transfer application, the throughput is calculated by dividing the transferred file size (bits) to the download duration (seconds) to get the throughput in bits per second. When the users share the same network resources; due to dynamic changes in network conditions, throughput for real-time services may not be sufficient if all data streams get the same scheduling priority.

End to end delay is another criteria to fullfill QoS requirements of each traffic flow because it might take a long time to reach its destination. The average end to end delay in second is calculated by dividing the sum of end to end delay of all the received packets to the number of received packet. The lost packets are excluded in the average end-to-end delay calculation.

The network performance of a user is often measured not only in terms of delay, but also in terms of the packet loss which is measured as the percentage of the number of received packets to the number of the total sent packets. A lost packet is retransmitted for TCP traffic to ensure that all the TCP data are reliably transferred from source to destination.

A fair distribution of system resources among different users is another important QoS parameter. There are several mathematical definitions of fairness. Our fairness measurement depends on the Jain's fairness index. Equation 3.4 shows the calculation where the fairness index changes between zero and one. When all the users have the same throughput, the fairness index becomes one that indicates the ideal fairness scenario.

5. SIMULATION RESULTS

In this section, the simulation environments of the proposed algorithms are explained and simulation results are shown followingly. A brief description of the simulation tool OMNeT++ is also included in the simulation environment part.

5.1 Simulation Environment

5.1.1 Simulation tool OMNeT++

OMNeT++ is an open source discrete event simulator which is especially designed to support simulation of network systems. It is mostly useful to build and represent hierarchical structures that are frequently used in design environment. The platform of this simulator is mainly implemented using the C++ programming language. Also, since OMNeT++ uses GCC tool chain or the Microsoft Visual C++ compiler, it can be executed in three most common platforms: Windows, OS X, Linux [33].

OMNeT++ can be operated freely for non-profit use and provides a powerful and consistent open-source library that can easily be used by research-oriented institutions. It is mainly used in research fields especially in communication networks area. Another advantage of this tool is different model structures can be implemented in OMNeT++ environment although these models are built outside of the OMNeT++ platform. Moreover, specific models and frameworks such as VANET [32] and INET [34] and can be used to extend the functionality provided by OMNeT++ platform.

OMNeT++ is designed to simplify the simulation of network systems [35]. It provides visualization and customization with its integrated development environment (IDE) [39]. OMNeT++ IDE contains multiple source code editors related to different activities. Also, IDE provides single environment to gather multiple structures in a single compound and compile them into a single unit. What is more, network component modification can be completed without coding by OMNeT++ IDE tool. The simulation environment is shown in Figure 5.1.

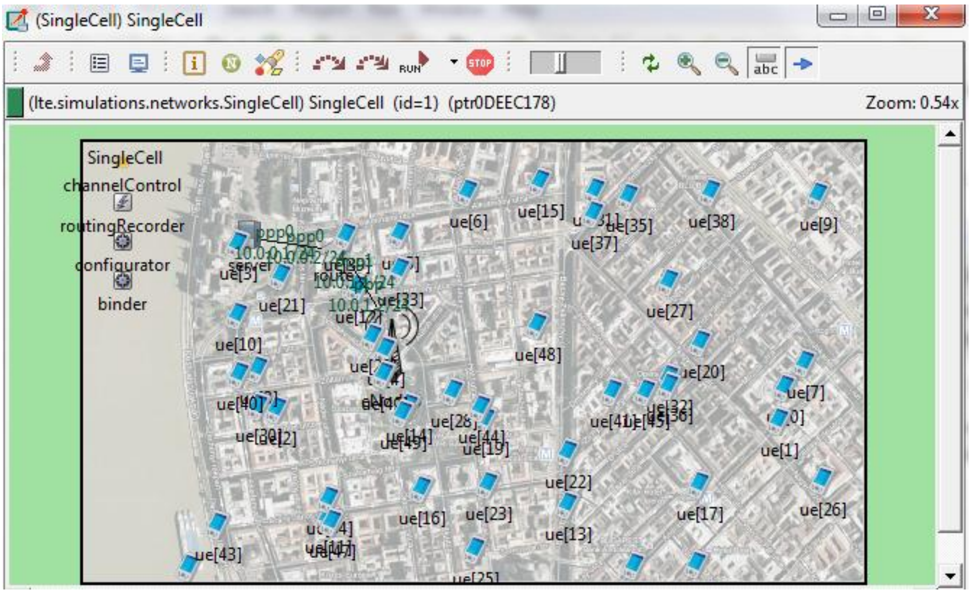

Figure 5.1 : OMNeT++ Simulation Environment Example.

OMNeT++ simulation tool has two kinds of user interfaces. They are TKENV and CMDENV interfaces. TKENV is OMNeT++'s GUI as shown in Figure 5.2.

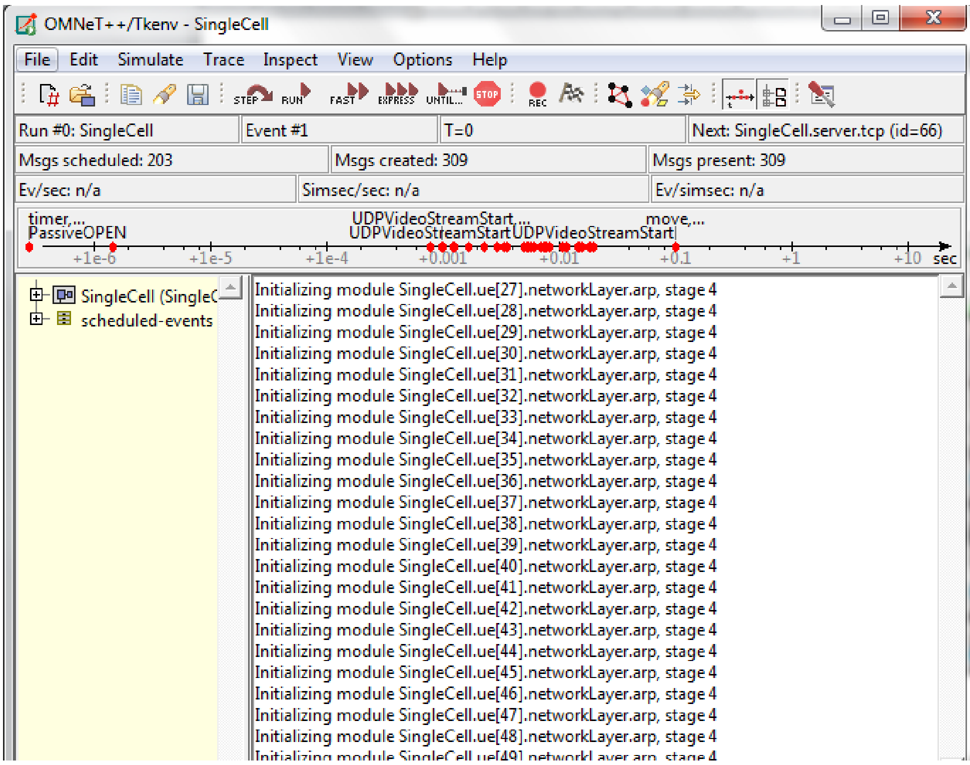

Figure 5.2 : OMNeT++ Simulator GUI.

It is built up to show automatic animation, module output windows and object inspectors. Automatic animation in OMNeT++ is capable of animating the owner of messages on network charts. Module output windows make it possible to open separate windows for the output of individual modules or module groups. Object inspector, which can be used to display the state or contents of an object in most appropriate way, is a GUI window associated with a simulation object.

In this thesis OMNeT++ simulation environment is used with the help of “simulte-master” module, is open resource developed to simulate LTE access network, to analyze and simulate proposed scheduling algorithms.

5.1.2 Simulation scenarios and parameters

As shown in Figure 5.3, the scenario involves a server, one eNodeB and several different types of UEs.

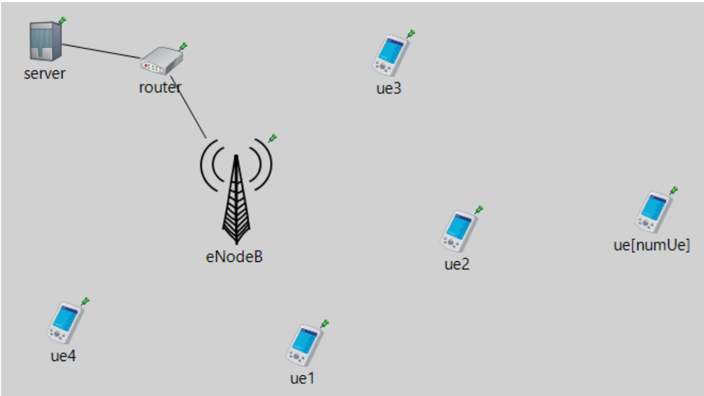

Figure 5.3 : Simulation Environment.

A brief illustration of the simulation scenario is presented in Table 5.1.

Table 5.1 : Simulation Parameters.

Parameters	*Values*
eNodeB	Single Cell Base Station
User Equipment	Min No. = 50; Max No.=150; Mobility = StationaryMobility
Physical Configuration	Number of RB = 100 Number of Subcarrier= 12 OFDM symbol = 7 Bandwidth = 20 Mhz
Path Loss Model	URBAN Macrocell
Traffic Model	Traffic Type= UDP video,VoIP and TCP (FTP) Packet Size = 1000 Byte Sent Interval = 10 ms

As can be seen Table 5.1, simulation parameters are demonstrated. SISO (Single Input Single Output) system with Single Cell configuration is used during simulations. While UDP traffic is represented downloading video from server and VoIP communication, TCP traffic is designed for FTP users. Mobility of the users is chosen as Stationary Mobility in 20Mhz bandwith that provides 100 available resource blocks to allocate between mobile users.

For each proposed algorithm simulation environment uses the same simulation paramaters. The test condition is performed by changing number of RT and NRT users.

5.2 Simulation Results of Algorithms

5.2.1 Algorithm_1 and Algorithm_2

In the first set of experiments, the performances of Algorithm_1 and MaxC/I are compared by varying the number of UEs from 60 to 150 shown in Figure 5.4. For all experiments, 80% of total users have only real-time traffic (UDP) while 20% of total users have only nonreal-time traffic (TCP). It means, for example 120 UEs with real-time traffic, 30 UEs with nonreal-time traffic for 150 UEs.

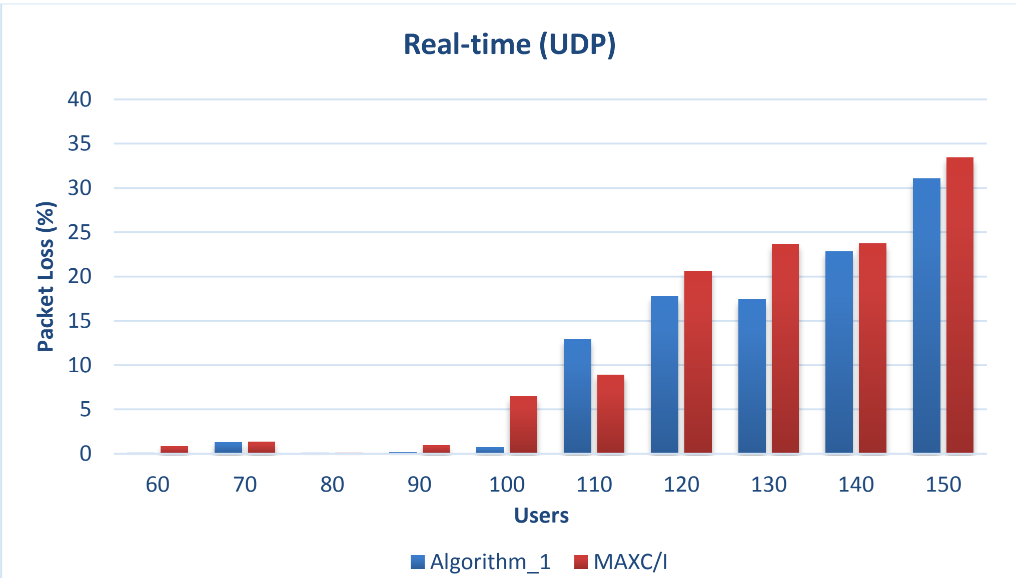

Figure 5.4 : Packet Loss of RT traffic (UDP).

In the second set of experiments, the number of UEs is set to 150 and the effects of the traffic prioritization parameter on the performances of real-time and nonreal-time users are analyzed by varying X between 10% and 40%. In other words, the simulations of Algorithm_2 are realized only with 150 users by changing the percentage of TCP traffic which is given higher priority. This scenario is particularly selected to demonstrate the effects of the reconfigurable traffic prioritization mechanism. Since there are only 100 RBs, radio resources are not sufficient to support 150 UEs.

Figure 5.4 demonstrates the packet loss percentages for real-time (UDP) traffic as the number of UEs varies from 60 to 150. Each UDP flow has constant bit rate traffic (packet size of 1000 bytes and send interval of 10 ms). The amount of traffic injected to the network increases when the number of UEs increases. Since the number of resource blocks is limited to 100, the packet loss rates are very low when the number of UEs is less than 100 for both algorithms. Algorithm_1 provides higher priority to the real-time flows when the resource blocks are allocated to UEs, and hence the average packet loss rates are lower compared to MaxC/I when the number of UEs is higher than or equal to 100. Followingly, Figure 5.5 presents the throughput results for nonreal-time (TCP) traffic.

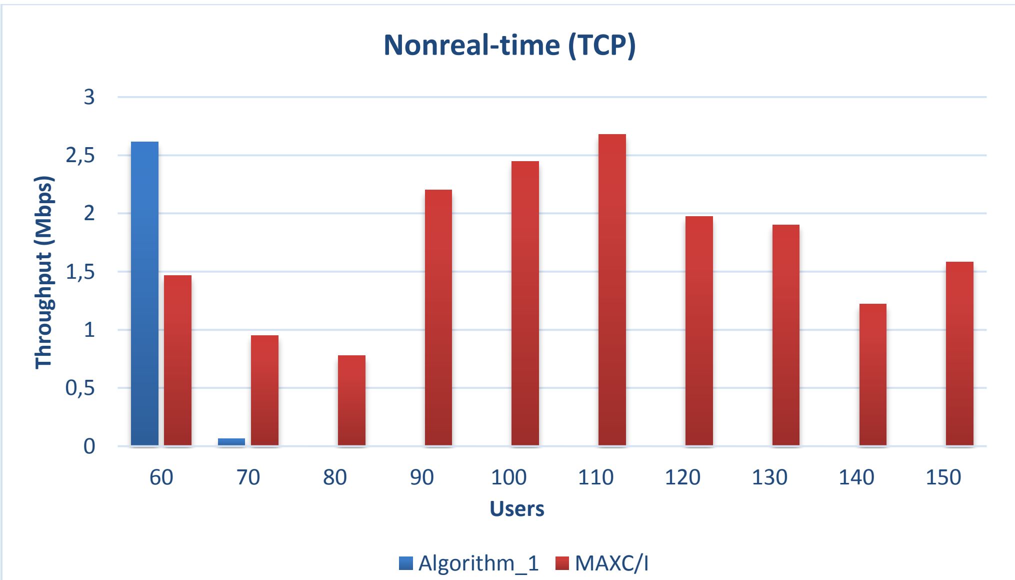

Figure 5.5 : Packet Loss of NRT traffic (TCP).

Since Algorithm_1 provides strict priority to real-time flows, throughput values for nonreal-time traffic become negligible when the number of UEs is beyond 80. However, MaxC/I always provides certain throughput values irrespective of how many UEs are in the system.

Figure 5.6 and Figure 5.7 show throughput values for real-time traffic and average packet loss rates for real-time traffic, respectively, when Algorithm_2 is used. The x-axis represents the percentage of nonreal-time traffic users whose priorities are made equal to real-time traffic users (the parameter X in Algorithm_2, where X% of nonreal-time flows are moved towards the top of the scheduling list). For these experiments, the number of UEs is set to 150. The numerical values in Figure 5.6 and Figure 5.7 are used for MaxC/I and Algorithm_1 as reference (the results do not depend on X).

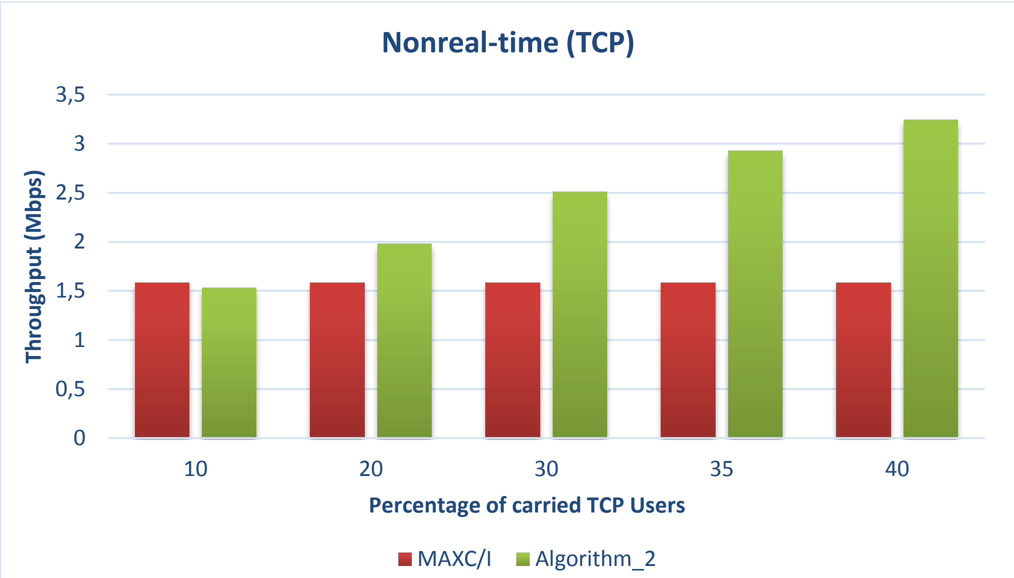

Figure 5.6 : Throughput of NRT traffic (TCP).

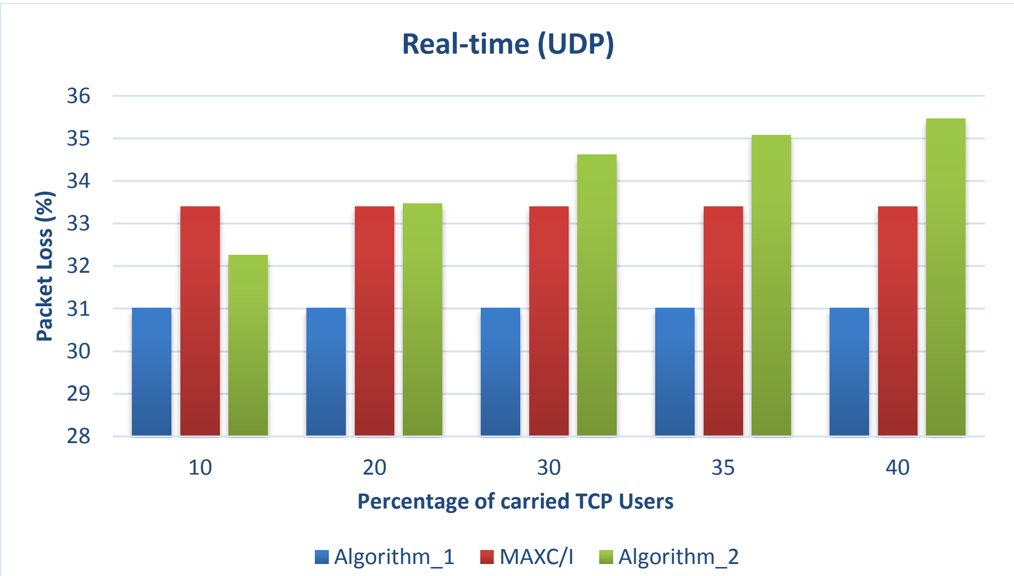

Figure 5.7 : Packet Loss of RT traffic (UDP).

As depicted in Figure 5.6, when the value of X increases, the throughput values of nonreal-time traffic increase. However, these improvements reflect performance degradations in the average packet loss rates for the real-time users as shown in Figure 5.7. These results demonstrate that the tradeoff between the performance improvement of UDP flows and the performance degradation of TCP flows can be controlled by tuning the reconfigurable traffic prioritization parameter X of the proposed scheduling algorithm. Accordingly, Figure 5.8 shows fairness results for nonreal-time traffic where 5.9 displays the fairness of the real-time traffic.

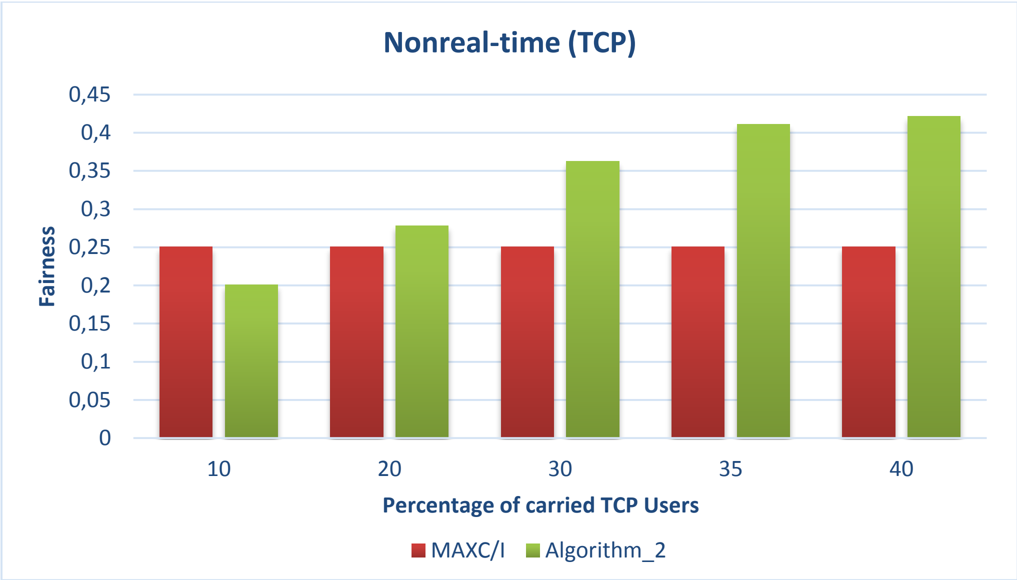

Figure 5.8 : Fairness of NRT traffic (UDP).

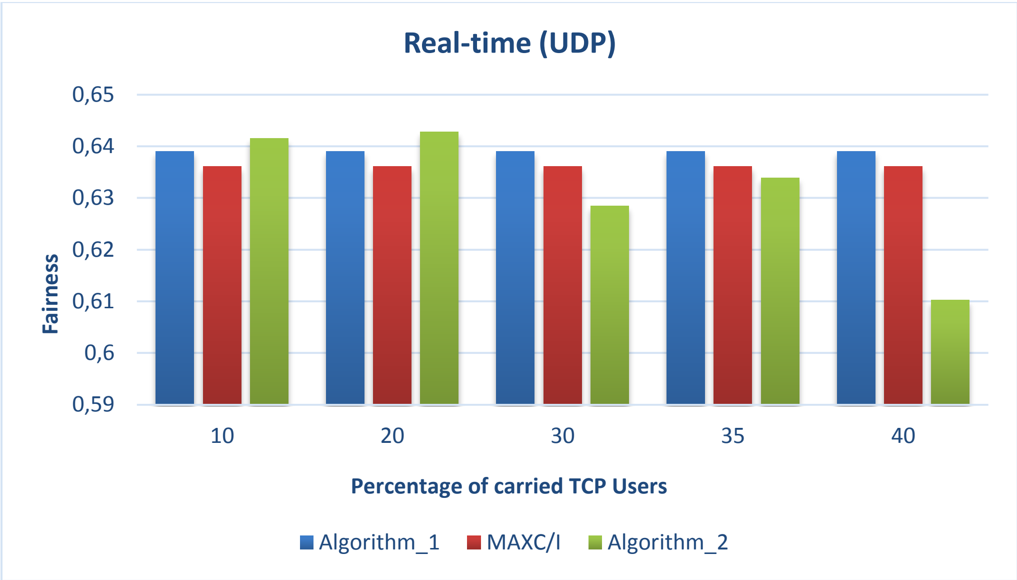

Figure 5.9 : Fairness of RT traffic (UDP).

The simulation results confirmed that when the parameter X increases, Algorithm_2 yields more fair results for nonreal-time traffic. Conversely, real-time traffic fairness is affected by this tradeoff that after the 20% it drops eventually.

5.2.2 Algorithm_3

In this set of experiments, the performances of Algorithm_3 is compared with MaxC/I and PF by varying the number of UEs from 60 to 150. For all experiments, 80% of total users have only real-time traffic (UDP) that are %40 VoIP traffic and %40 video streaming while 20% of total users have only nonreal-time traffic (TCP). It means, for example 40 UEs with real-time VoIP traffic, 40 UEs with real-time video streaming traffic, 20 UEs with nonreal-time FTP traffic for 100 UEs.

The delay performance of the algorithms MaxC/I and PF are illustrated in Figure 5.10 and 5.11 in terms of VoIP and video traffic users among growing number of users from 50 to 150. Besides, 5.12 represents VoIP packetloss results.

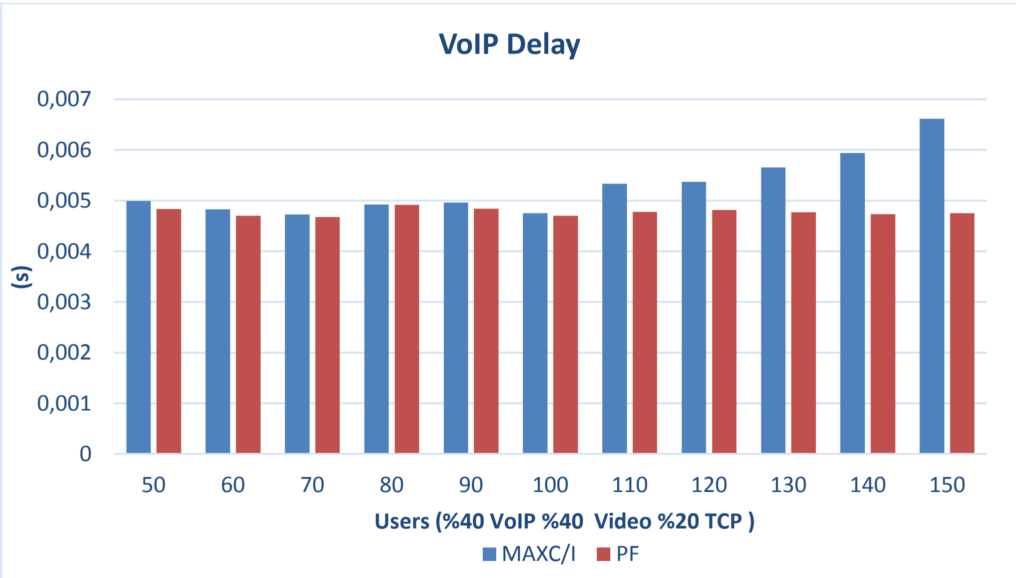

Figure 5.10 : VoIP Delay for MaxC/I and PF.

Delay distribution of the VoIP is achieved until the network loaded with 110 users. Neverthless, delay rate increases dramatically as it reaches the 150 users. In addition, video delay of the existing algorithms rises remarkably especially the network is used by 150 users.

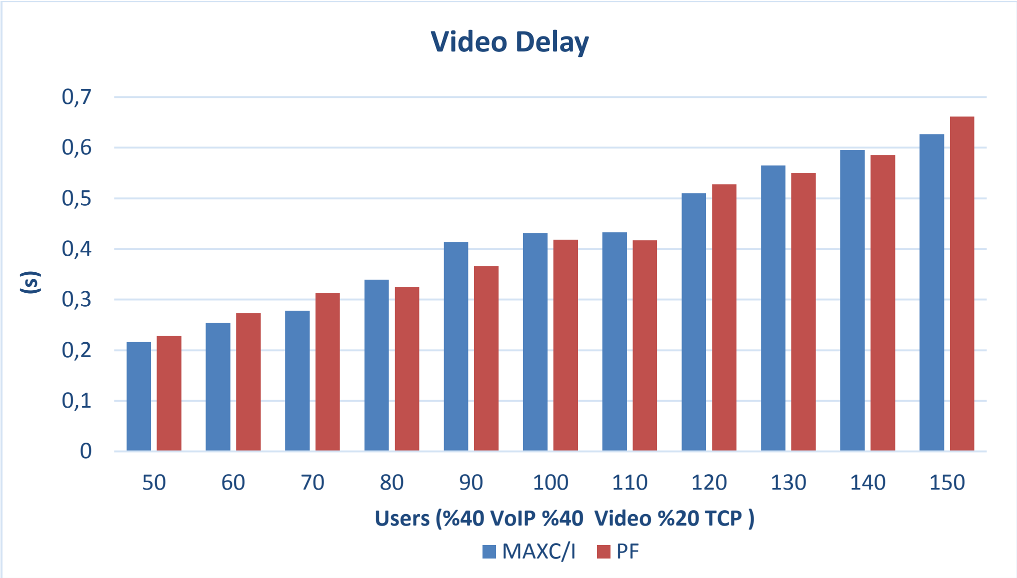

Figure 5.11 : Video Delay MaxC/I and PF.

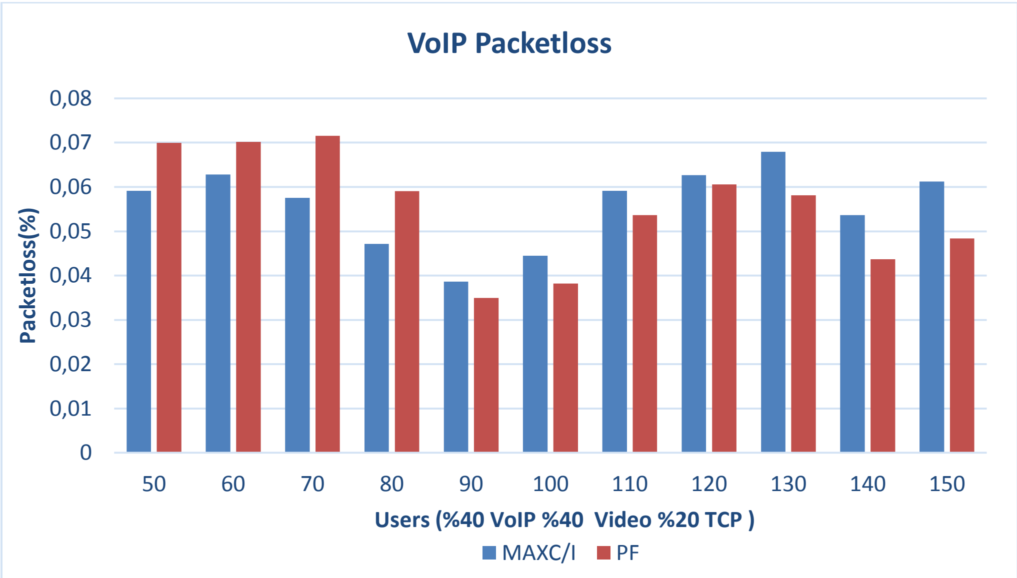

Figure 5.12 : VoIP Packet Loss for MaxC/I and PF.

Fig 5.12 shows the percentage of VoIP packetloss with respect ro MaxC/I and PF algorithms by changing number of users in ascending order between 50 and 150. The user diversity makes sense when the number of users exceeds 90. That point states the threshold where saturation of the number of available resource blocks starts. It can be seen from the graph that less than 90 users, MaxC/I perform better whereas PF algorithm works with lower packetloss above 90. Besides, after the user 130 the gap of packetloss between the algorithms are increased.

In Figure 5.13, it is evident that TCP packetloss among the users are distributed in favor of PF algorithm with lower percentages of packetloss. Conversely, MaxC/I packetloss is greater than PF for each network load condition.

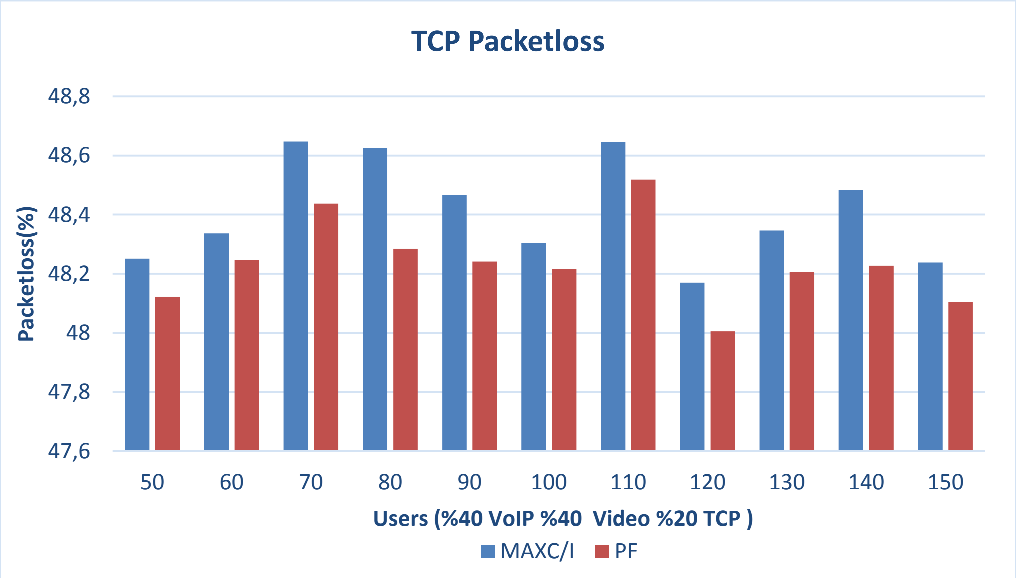

Figure 5.13 : TCP (FTP) Packetloss for MaxC/I and PF.

Figure 5.14 indicates the real-time traffic throughput with respect to MaxC/I and PF algorithms between 50 to 150 number of users.It is inevitable from the figure that real-time throughput increases with the growing number of users.

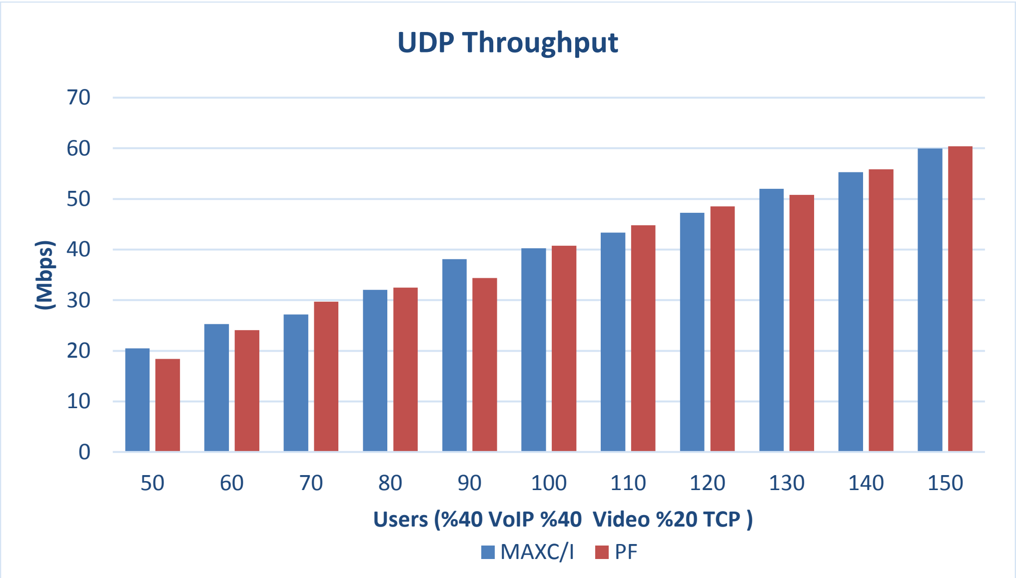

Figure 5.14 : UDP (VoIP,Video) Throughput for MaxC/I and PF.

Figures 5.15 illustrate the throughput performance of the real-time and nonreal-time traffic according to MaxC/I and PF algorithms. As it is expected throughput rates are increasing as a result of increasing amount of user.

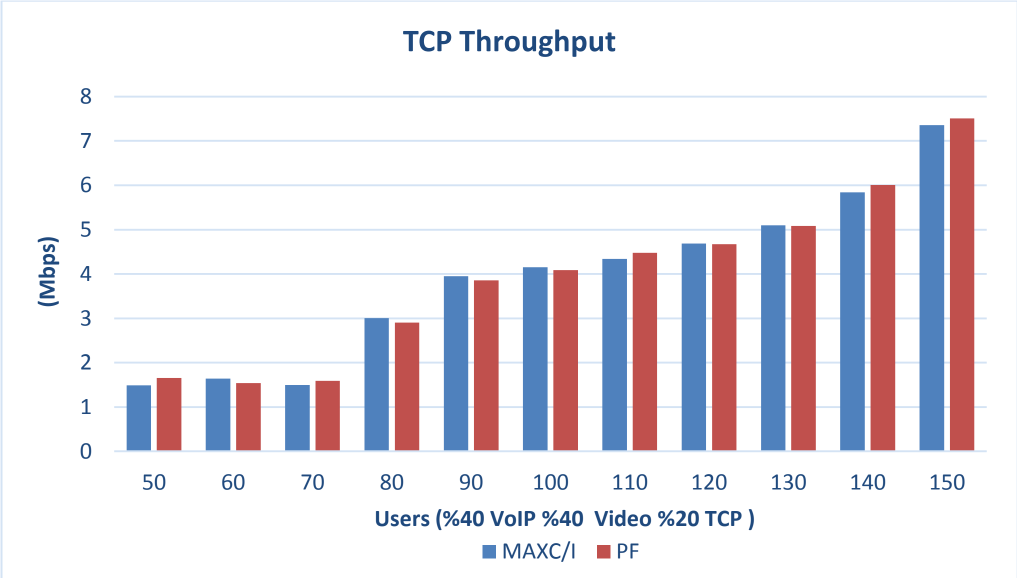

Figure 5.15 : TCP (FTP) Throughput for MaxC/I and PF.

Figure 5.16, 5.17 demonstrates the delay performance of the proposed algorithm, Algorithm_3, according to MaxC/I and PF algorithms. Vertical values show how many nonreal-time users are moved to realtime part of the list.

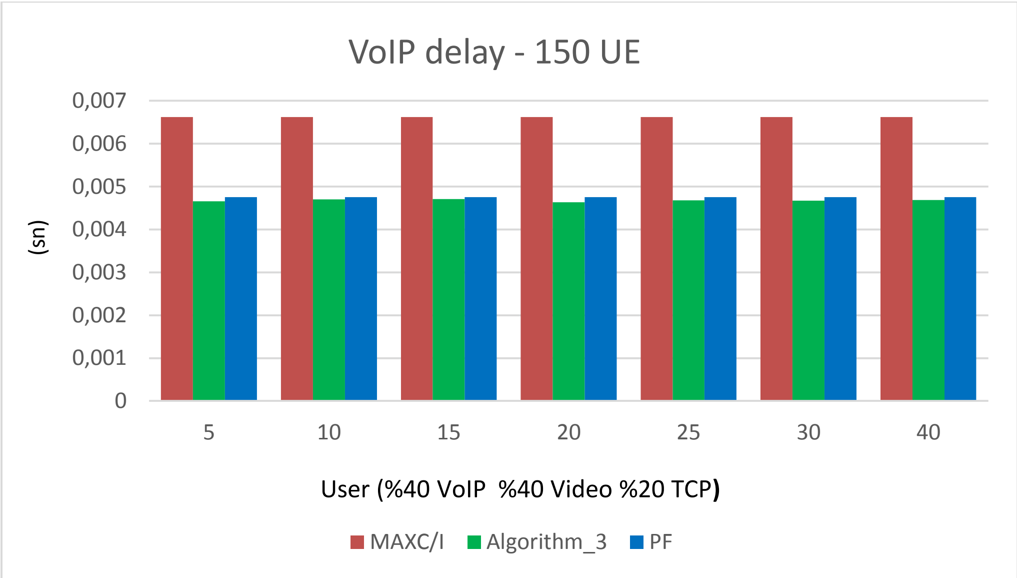

Figure 5.16 : VoIP delay for MaxC/I,PF and Algorithm_3.

This percentile method is used to define some thresholds in order to fulfill the QoS requirements between real-time and nonreal-time traffics. It can be concluded from the graphs that proposed algorithm can achieve the average delay for VoIP even much better than MaxC/I algorithm. However, video delay is going up dramatically after 20% in an unfavorable way when it is compared to MaxC/I. On the other hand, this delay does not exceed the values of PF algorithm even for 40%.

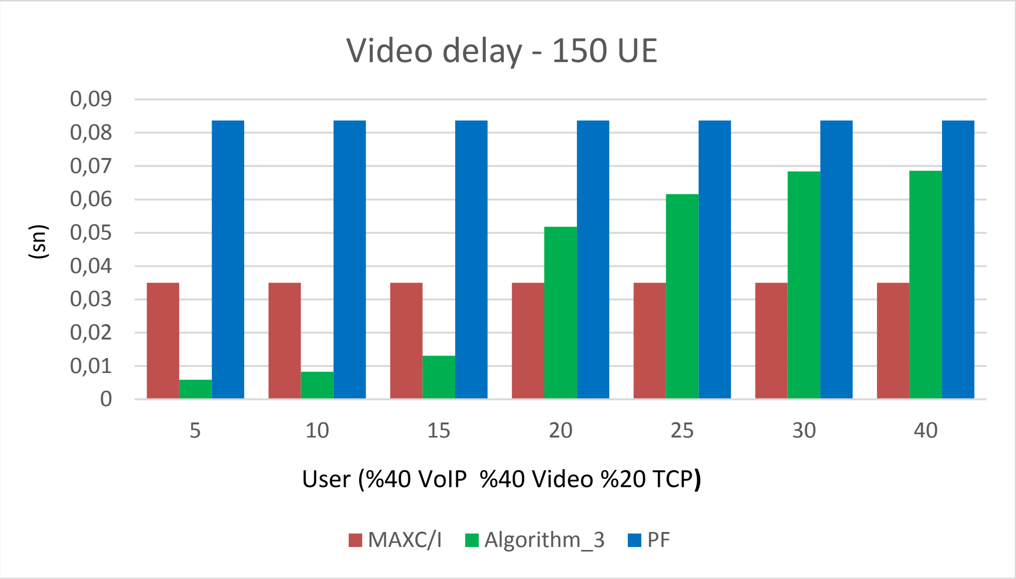

Figure 5.17 : Video delay for MaxC/I,PF and Algorithm_3.

Moreover, it can be seen from the charts that PF algorithm succeeds in minimize the VoIP delay where MAXC/I perform well for Video delay. Reconfigurable parameter is required to accomplish the quality service by Algorithm_3. Consequently, Figure 5.18, 5.19, 5.20 indicate TCP, Video and VoIP throughput results respectively.

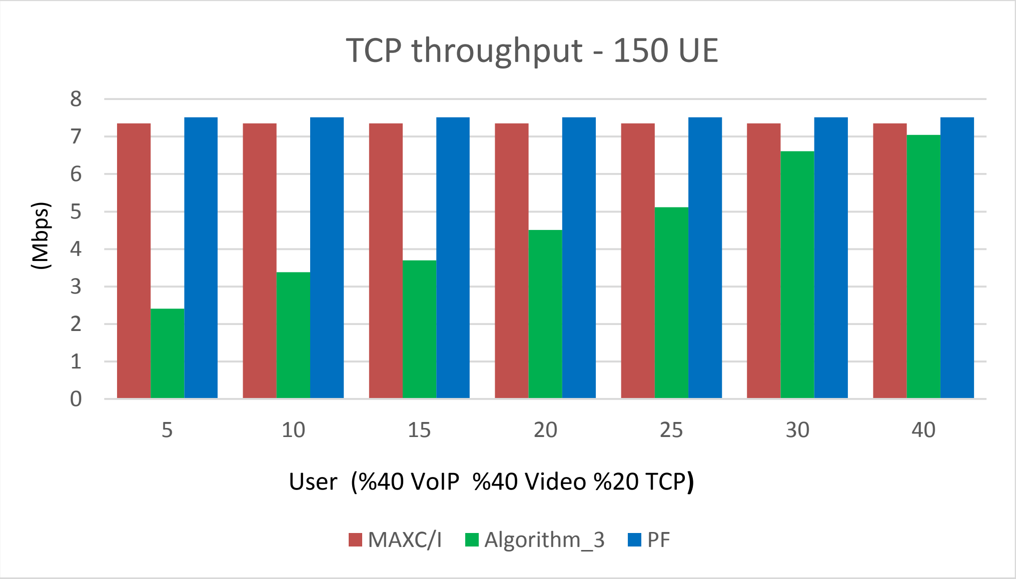

Figure 5.18 : TCP Throughput for MaxC/I,PF and Algorithm_3.

It can be obtain from the results that TCP throughput value is growing because percentage of the nonreal-time user increases relatively in the candidate list. Algorithm_3 supports the stability between MAXC/I and PF algorithms until it reaches the 40%.

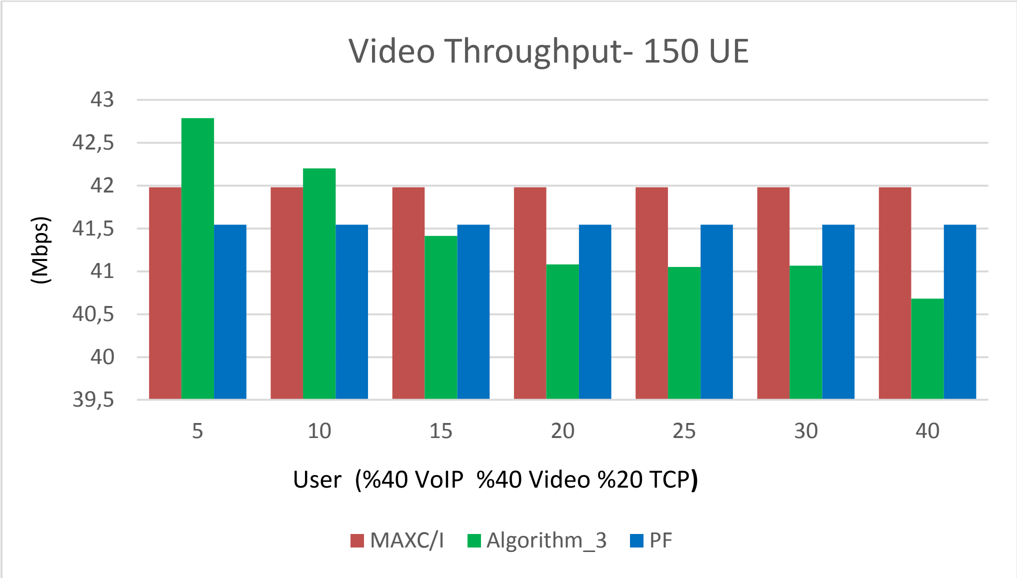

Figure 5.19 : Video Throughput for MaxC/I,PF and Algorithm_3.

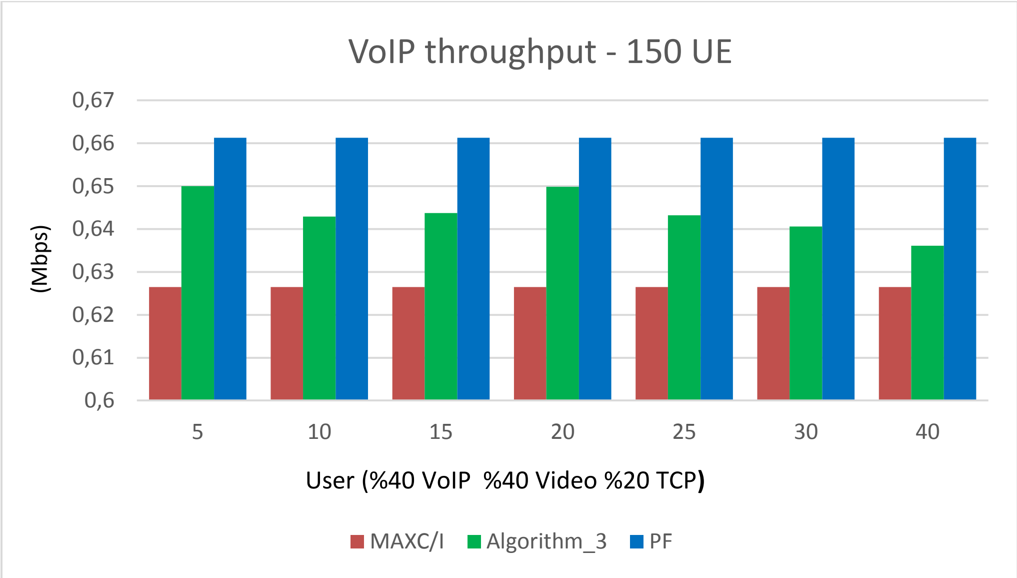

Figure 5.20 : VoIP Throughput for MaxC/I,PF and Algorithm_3.

Video throughput and delay graphs 5.19, 5.17 state the tradeoff the prioritization of nonreal-time users. Since MAXC/I and PF algorithms firstly consider the VoIP traffic users, Algorithm_3 move VoIP users at the top of the list that gives an advantage to keep performance expectation between PF and MAXC/I, Fig 5.20.

On the other hand, the tradeoff between delay and throughput can be observed especially after 20%. When the percentage of moving nonreal-time users is increased, TCP throughput values of them increase as well. Especially after 20% throughput is rising significantly. However, these improvements reflect the average packet loss ratio in real-time users. The average packet loss ratio in real-time traffic is going up dramatically after %20 in an unfavorable way when it is compared to MAXC/I,Fig 5.21. Moreover, delay and throughput results of video traffic are affected in that vein, Fig 5.17, 5,19.

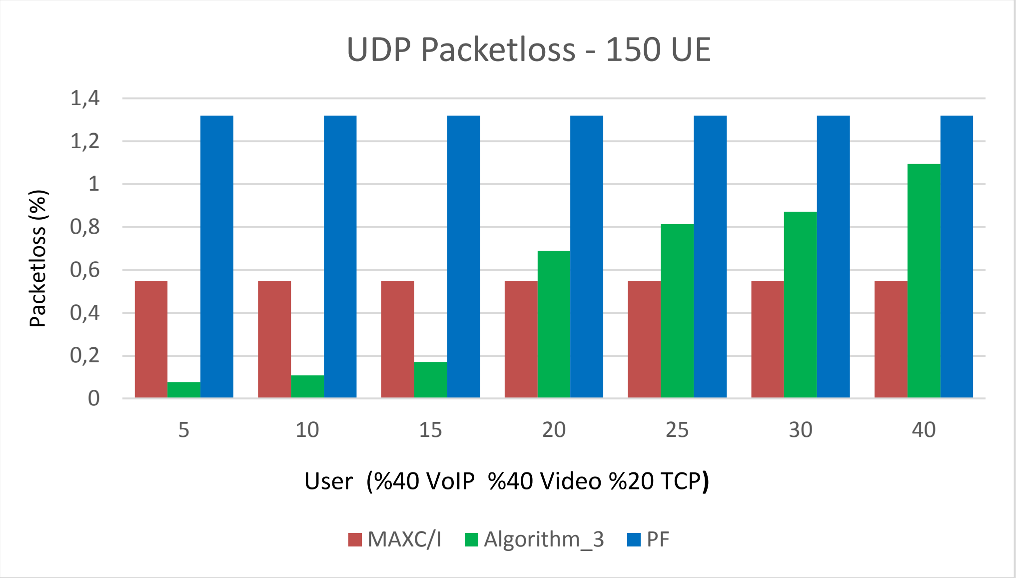

Figure 5.21 : UDP Packet Loss for MaxC/I,PF and Algorithm_3.

The graph of 5.22, 5.23 point out the fairness advantage of the proposed algorithm.

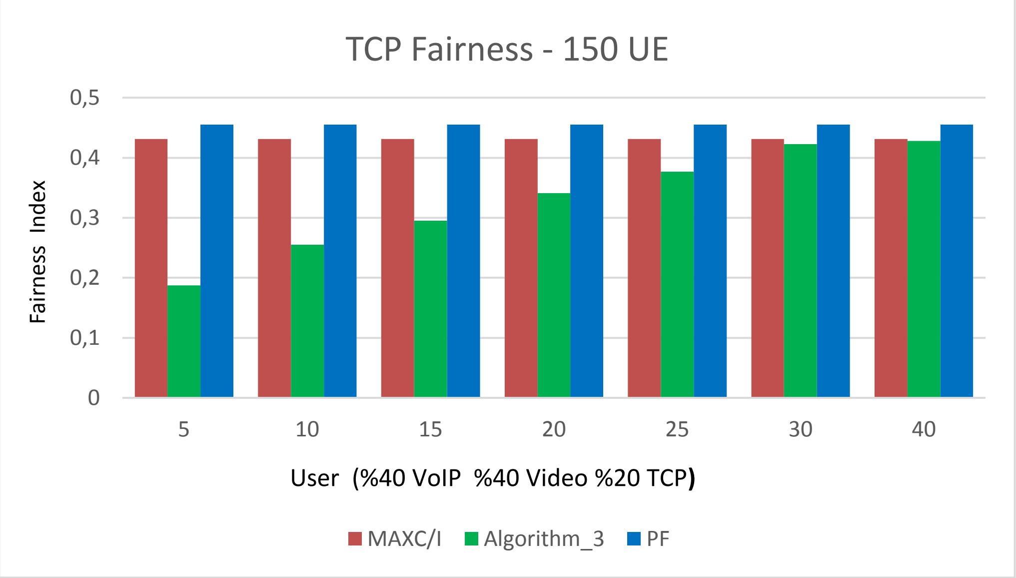

Figure 5.22 : TCP Fairness for MaxC/I,PF and Algorithm_3.

Although TCP fairness increase with respect to increasing number of TCP users, UDP traffic fairness can be maintained within reasonable interval between MAXC/I and PF algorithms.

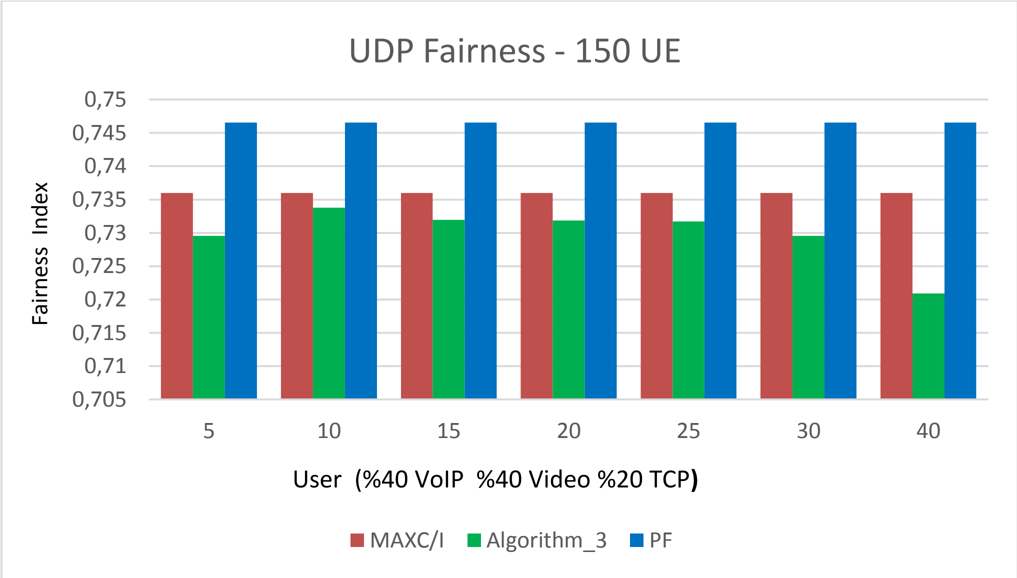

Figure 5.23 : UDP Fairness for MaxC/I,PF and Algorithm_3.

The fairness index parameter depicted in 5.22 and Figure 5.23 are illustrated to show TCP and UDP flows, respectively. This parameter shows how the system is serving users fairly by assigning a fair share of system resources. Hence, the figures show that the proposed scheduler achieves better fairness for different types of services.

6. CONCLUSION AND FUTURE WORK

The main reason behind the world's switching to LTE technology is to fullfill growing number of mobile user necessities such as higher end-user data transfer rates, lower latency for real-time services with lower cost of operation. Compared to previous networks, LTE introduces as a flexible radio interface to provide interopabilitiy of heterogenous networks, flexible bandwidth usage and support end to end IP based services. To objective these specifications LTE scheduler is the key enabler to ensure QoS requirements by maximizing the usage of limited resources.

In LTE; scheduling, radio resource allocation task, is carried out within eNodeB instead of controllers such as BSC or RNC in the traditional cellular network architecture. In this context, channel-dependent scheduling has a crucial role by allocating resources based on channel conditions. It performs how to share available radio resources between different terminals to achieve efficient resource utilization as possible. By periodically receiving the feedback information from the terminals, the scheduler in the base station dynamically adjusts scheduling decisions to maximize resource allocation and meet quality of service requirements.

In this thesis; QoS-aware reconfigurable scheduling algorithms which provide higher priority to the real-time users are proposed using channel-dependent scheduling. None of the existing scheduling algorithms such as Round Robin, MaxC/I and Proportional Fair do not take both the traffic types and channel qualities into account and there is no service guarantee for real-time traffic. However, our proposed scheduler takes both channel qualities and the traffic types into consideration and reconfigurable parameter gives an opportunity to the administrator to control the tradeoff between the performance of real-time and non-real-time traffic.

The main idea of the our algorithms is to meet the delay requirements of real-time traffic first and then maximize the performance of nonreal-time users in terms of their throughput, average packet loss ratio and fairness. Algorithms are appiled to minimize the amount of resources needed per user while still satisfying the QoS. In this regards, three scheduler algorithms are studied that from first to third, the algorithms are extended to reach more realistic results. First, incoming traffic is classified into real-time and nonreal-time flows which are inserted into a priority list according to traffic types and channel qualities. Algorithm_1 mainly consider the taking traffic types into accout that classifies real-time and nonreal-time traffics to prioritize real-time flows. The scheduler assigns available resource blocks to user equipments (UEs) using the order in the list. Algorithm_2 focuses on reconfigurable parameter that gives a control to the network administrator between real-time and nonreal-time flows for efficient resource allocation. A reconfigurable parameter in the proposed scheduler determines the resource prioritization level for real-time services, and hence allows network administrators to more effectively control the resource allocation according to the QoS requirements of UEs and changing network dynamics. Algorithm_3 which is the extended version of Algorithm_2, considers both channel qualities and multi-traffic types of terminals into account to support QoS requirements the user perceived. Moreover, a reconfigurable parameter is included to achieve the resource prioritization level for real-time services, and hence allows network administrators to more effectively control the resource allocation according to the QoS requirements of users and changing network dynamics. In this way, by considering channel qualities and traffic types to support QoS and minimize human involvement in network operation with reconfigurable parameter satisfies the self-organizing network requirements.

OMNeT++ based simulation results demonstrate that the tradeoff between the performance improvement of real-time flows and the performance degradation of nonreal-time flows can be controlled by tuning the reconfigurable traffic prioritization parameter of the proposed scheduling algorithm with multiple traffic scenarios.

For the future work, scenarios and simulation environment can be extended to support more realistic network dynamics (e.g., multiple base stations, realistic traffic scenarios including more traffic classes). To add, users with various mobility patterns will be introduced. Furhermore, the proposed algorithm can be extended to run autonomously without human in the loop by receiving certain performance results (e.g., delay, packet loss, etc.) as feedback from the terminals instantanously. Moreover, autonomous operation of the scheduling will decrease the operational cost substantially and enable an important SON feature in LTE networks.

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APPENDICES

APPENDIX A: VoIP – Streaming – FTP

VOIP:

VOIP stands for ‘Voice Over Internet Protocol’ and implies multimedia communication over Internet Protocol. Voice is digitized, compressed and packetized, and is sent over IP network in this technology. Thus, it is fully packet-switched technology. Codecs are used to encode voice. Some popular codecs are G.711, G.722, G.729, AMR etc. In 3GPP, Adaptive Multi Rate (AMR) wideband and narrowband codes are used. Different codes have different encoding rate. For example, G.711 converts voice into 64 Kbps stream, while G.729 converts voice into 8 Kbps stream VoIP uses Session Initiation Protocol (SIP) or H.323 as signaling protocol to initiate and terminate the connection between voip users. Generally, VOIP services use User Datagram Protocol (UDP) as datagram protocol as these services are delay-sensitive. UDP does not guarantee the delivery of the packets as there is no retransmission scheme. However, this is simple, faster and efficient scheme. Real-time Transport Protocol (RTP) is data transfer protocol which deals with the transfer of real-time data. RTP is used to maintain Quality of Service (QoS). RTP works only on UDP. VOIP traffic consists of talk-spurts and silent periods. One VOIP packet arrives every 20 msec during talk-spurts, while one Silence Indicator (SID) packet arrives every 160 msec during silent period that VOIP is peer-to-peer technology. Skype is an example of VOIP application.

Streaming:

Streaming generally means the process of sending digitized video and audio packets over internet so that the end user can watch video or hear audio in real-time fashion. With this technology, end-user can play file (video, audio) before entire file is downloaded. Unlike VOIP, streaming supports multicast technology, that is, many users can watch/hear the single stream simultaneously. The generally used audio codec in this application is MP3, while H.264 and MPEG-4 are used as video codec Session Description Protocol (SDP) is used as signaling protocol used in describing multimedia sessions like session initiation, session invitation, etc. Other protocols used in this application are RTP, RTP Control Protocol (RTCP), Hyper Text Transfer Protocol (HTTP), and Real-time Streaming Protocol (RTSP). RTCP is a control protocol and works in conjunction with RTP. RTSP is application-level protocol used in transferring real-time data. RTSP allows user to play the downloaded and decompressed packets, while it continues downloading and decompressing next arrived packets in parallel. Thus, it enables end-user to use real-time data. Streaming takes help from another application-layer protocol HTTP as well. To begin streaming, user has to go to web-page. The web-page then directs to RTSP server.

FTP:

FTP is an application-layer based protocol which allows users to transfer files between two computers over IP network. It supports Transmission Control Protocol (TCP) only as transport layer protocol. TCP is connection oriented transport layer protocol which takes consideration of retransmission for error control, and thus guarantees successful delivery of data. TCP also supports flow control, that is, window size to transfer data is already negotiated between sender and user. This scheme ensures that user gets data only at that data-rate that it supports. FTP works with server and client approach. When client sends a file-transfer request to server, TCP connection is established. TCP then segments the data and put unique identification to each segment. For each segment, client and server do communications with acknowledgements. If any segment is lost or corrupted, that segment is retransmitted. Some commercial software such as 'FileZilla', 'CoreFTP' can be used as FTP client and server. In this project, TI built-in FTP client has been used.

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