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Study and Performance Analysis of LTE MAC Schedulers for M2M

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Master of Science in Electronics
Submission date: February 2016
Supervisor: Kimmo Kansanen, IET
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Abstract

Cellular systems are forecasted to play a fundamental role in the future Machine-to-Machine scenario. The 3GPP LTE networks appear to be the de facto standard for machine type communications. Beside opportunities given by M2M devices spreading, such as vehicular-to-vehicular communications and environmental monitoring, the operators will have to deal with a greater number of devices connected, which don't fulfil with the nowadays human centred traffic standard characteristics and will force to re-standardize the current uplink scheduling procedures.

In this report an extensive study of the literature papers, both standards and M2M centred, is carried on a dense machine scenario. Problems such as Human-to-Human throughput drops are high lined with the raise of the number of M2M devices present. Moreover the channel utilization, both in uplink and in downlink, drops drastically among the studied scheduler schemes.

A new M2M Aware Scheduler, with the goal of maximise the medium utilization, is designed and implemented. Its results over the simulations shows that M2M and Human-to-Human devices could live over the same LTE Network but new schedulers, such as the one here presented, have to be further studied and analysed.

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1. Motivations

The growing and spreading of new technologies is now emerging in the everyday consumer's life and that leads in new challenges on the communication engineering with point. While new applications of IoT (Internet of things) devices are constantly released with the aim to make our homes safe, new fields of interest like Vehicular to Vehicular communication are deeply studied, that will take to the point of asking if the underlined communication infrastructure is enough to carry and to route those kind of new communication forms without interact with the "traditional" network users. From now on in this elaborated those new kind of communications will be labelled as Machine-to-Machine communication (or with the acronym M2M) and will be further analysed in its characteristics and peculiarity taking as a starting point the literature nowadays available. M2M communications are currently based on contemporary wireless communication technologies like Global System for Mobile communications/General Packet Radio Service (GSM/GPRS), which are fulfilling the requirements of existing M2M applications sufficiently. The contemporary communication technologies offer low-cost deployment of M2M devices with convenient deployment and roaming facilities. The current main infrastructure used to carry M2M traffic is the second generation GPRS one, which was not designed with the aim of carrying not human traffic, will not be able to handle the amount of traffic generated for long. The trend which the hardware manufactures are following is to move the data exchange to the new Long Term Evolution, relate 8 of the 3GPP Third Generation network, to solve the traffic problem. The LTE, despite its name, was not initially designed to handle different data traffic from the human ones, but the nature of its packet switching core can be and will be leveraged to serve the M2M traffic as well.

The previous statement leads directly to the main goal of this elaborated: The analysis of study of the M2M traffic flow and its effect on the classic human users. Is thought that this analysis has a bold importance since the way of handling machine to machine traffic is as a not deferrable mile stone for network carrier operator.

Network operator are facing the challenge of the serving their usual, human traffic focused, clients and ensure them a reliable channel while on the other hand handle the communication from machine type devices which can deteriorate the human channel. To study that in this elaborated will be provided a study analysis on both type of traffics. In a prelaminar analysis it’s easy to underline as the machine traffic is not busty at all: it’s characterized by small packet as payload and a sending rate in the scale of one packet each hour/day. More over the traffic is mainly uplink related, characterized by small size payload packets, since the nature of the machine to machine interaction which follows the paradigm: analyse the environment, report data to the network sink and go in sleep mode to preserve battery.

Those behaviour could lead to think that this kind of impact on a wide network could be easily neglected but the trends [1] says clearly that in the nearly future we will have a huge density of machine type devices in the order of hundreds or thousands of devices for base station area. The only way to prevent a data exchange overflow, coming from the machine nodes, is to shape the decision of allowing the channel to the nodes. The traffic shaping and filtering can be easily accepted in the sense that different type of devices have different Time To Live of the message, that means that while a H2H (human to human) communication has to be served with a delay of 1ms (in the case of voice communications), M2M one can have a Time to Live of minutes or days. That’s allows the network maintainer to plan how a Machine to Machine aware Scheduler should work.

So in this elaborate the arising problem is tried to be addressed by the LTE Scheduler view point, first analysing the literature schedulers and then developing and testing a new M2M Aware Scheduler. On the transport layer will be utilized the TCP/IP protocol which ensure a reliable communication end to end both for LTE human devices and LTE M2M devices. All the simulation provided in this elaborate will be performed through the NS-3 simulator, which has its LTE module designed from the CTTC of Barcelona [2]. The LTE module implements the 3GPP release 8.

2. Traffic characterization

2.1. Machine Traffic Communication scenario

Following the rise of the variety and number of devices, also the mobile data traffic will rise significantly in the future [1]. As the “Internet of Things” is a new paradigm for devices that are becoming connected, M2M communication can be seen as an important part in this trend for the near future. Along the reason for an increasing interest in remote monitoring, such as health and environmental monitoring, the carrier focus point should be set on how to maintain a certain user network experience. Cellular broadband connectivity became cheaper and ubiquitous over the past years, and also the costs of devices with integrated sensors, network interfaces, are becoming smaller, cheaper and more powerful. That trend allows industries a wide range of applications and devices, and new areas where those applications can be deployed. The M2M applications are expected to offer a diverse range of services, including narrowband applications transmitting data infrequently, for that reason the first architectural problem arise: LTE is primarily developed for broadband data services. With narrowband applications, LTE cannot achieve spectrum and cost efficiency, as is shown in Performance metrics Chapter. Therefore, the integration of M2M communication, with their considerable lower data rates and smaller packet sizes, might have a considerable impact on the LTE system. The usage of LTE network resources by M2M devices can have a significant impact on the performance of regular LTE data traffic such as voice, video and file transfer.

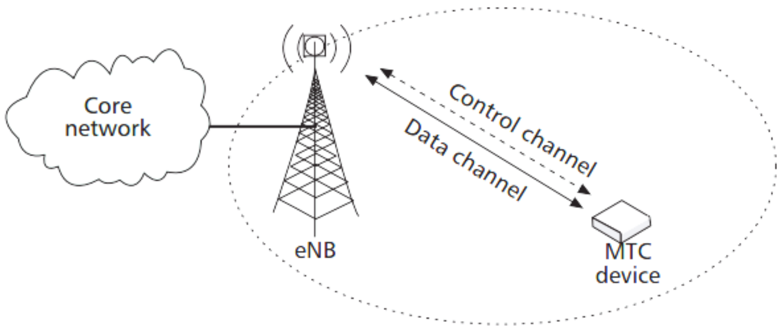

Figure 1: M2M Communication architecture

In 3GPP LTE at the moment is proposed that each MTC device attaches to the existing cellular infrastructure, Figure 1. The subsequent challenge thus lies on how to shape the management on the air interface access. To achieve successful M2M communications, quality-of-service (QoS) guarantees provisioning is the most important requirement to be investigate, due their different nature from the H2H communication. For MTC devices, some applications require deterministic timing constraints, and disasters occur if timing constraints are violated. For some other applications (e.g., gaming signals), statistical (soft) timing constraints can be acceptable. Different from applications in human-to-human communications (e.g., multimedia) that packet arrival periods typically range from 10ms to 40ms, packet arrival periods in M2M communications can range from 10ms to several minutes. Thus, how to effectively multiplex such massive accesses with enormously diverse QoS characteristics turns out to be the most challenging task.

2.2. Traffic models

As underlined in the previous paragraph the characteristics of M2M traffic is different to the existing, human-based network, traffic. Where human-based communication under comes to a certain session length, data volume and interaction frequency, M2M traffic follows some very specific traffic patterns. The most significant difference is the amount of data per transmitted packet: It is usually very small, i.e. only several (hundreds) bytes and refers to the nature of the generated data. As most of today's M2M devices are reporting sensor data like: temperature, humidity, energy usage the transmitted packets on "the air" consists of the measured data plus the corresponding protocol overhead, in this report the studied sensor will use the TCP protocol. In most cases, a device reports the measured data in periodical intervals to a remote end-point. Although those intervals range from several minutes to hours [3], the multitude of distributed M2M devices within a certain cellular network may create a considerable dense scenario. Following the [1] scenario, M2M traffic is studied to grow 22-fold from 2011 to 2016, which corresponds to an annual growth ratio of 86%. Although the most dominant devices will still be Smartphones and Personal Computers, such as laptops and notebooks, M2M devices will account for about 5% of the total mobile data traffic in 2016. Even though they will still play a rather small role in the entire mobile data traffic, the average M2M device will generate 266 MB of traffic per month. That value corresponds to a data volume of approx. 6 kB per minute.

Traffic modelling underline the design of a stochastic processes to match the behaviours the real measured data traffic [3]. Traffic models are usually classified as source and aggregated traffic models and aggregated traffic models (e.g., backbone networks, Internet, high-speed links).

Source Traffic Models	**Aggregated Traffic Models**
Data	Backbone networks
Video	Internet
Voice	High-speed links

Table 1: Traffic Model cases

MTC traffic fits into the aggregated traffic models class, since the typical use case includes numerous simple machines assigned to one server. This can be modelled as simple Poisson process, however, due to coordination (synchronizations) in MTC traffic, the respective arrival rate λ may be changing over time, λ (t) [4]. As long as the MTC devices become more complex (e.g., video surveillance), the more difficult becomes the approach of modelling them as aggregated traffic. The global data flow may exhibit high order statistical properties which are difficult to capture [5]. Is further expect this effect to be amplified by the synchronization of sources. Source traffic models which can capture the coordinated nature of MTC traffic are available in literature but they are designed for a low amount of sources and are too complex, nowadays, for MTC traffic.

Future networks are forecasted to support up to 30 000 MTC devices in one cell, which is orders of magnitude more than today's requirements [6]. Nowadays networks suffer serious Quality of Service (QoS) degradation if confronted with simultaneous access attempts from many devices. Study and understanding of this phenome is the main focus of this report. For multiple access and capacity evaluations, aggregated traffic models such as Poisson processes, are a satisfactory description of reality and therefore largely deployed.

The 3GPPP Model

The 3GPP model is made of two scenarios, Model 1 and Model 2. The first one describes the uncoordinated traffic and the second one the synchronous traffic. Both scenarios are defined by a distribution of packet arrivals over a given time period T, where the Probability Density Functions (PDFs).

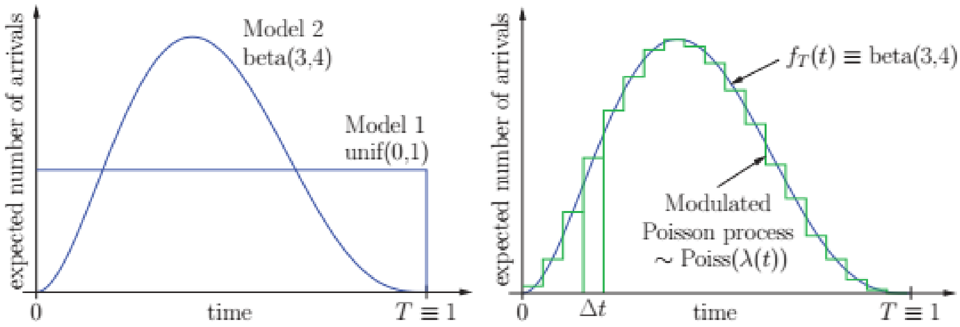

Figure 2: 3GPP Traffic Models

While the 3GPP Model 1 could be achieved by a uniform distribution (Figure 2 left) is here stated that the Model 2 can be seen as a Modulated Poisson Process. The 3GPP model reaches its limits for further requirements such as:

- few machines in the scenario, so that a data source has to be associated with a fixed location;
- multiple packets (bursts) coming from the same machine;
- the synchronous traffic (Model 2) influences the regular traffic (Model 1);
- the network has an influence on the traffic patterns.

2.3. Transmission Control Protocol

The Transmission Control Protocol (TCP) has the task to establish the communication between two network users, to ensure the reliability of the data transferred between them and to release and close the connection once done. TCP is capable to transfer a data flux in both the communication directions (full-duplex), managing when block or continue its operations. Since TCP haven't any bias on the underling hardware deployment, is possible to deploy it on a local Ethernet network or, has it become fundamental, on the internet.

Similarly to the User Datagram Protocol (UDP) the TCP is located in the stack layer model above the Internet Protocol Layer (IP), which handles the transfer and the routing of a single packet until its destination, but, unlike the UDP, TCP keeps traces of the sent packets in the order to restore the communication from a data loss through the path to the sink. As in the UDP protocol, in TCP is allowed the communication of different applications on the same machine in pipeline, it's achieved by de-multiplexing the packets traffic that feeds the applications thought the use of port numbers to identify the final communication sink inside the same machine. The main difference between from the UDP is that TCP grantees a reliable communication (Reliable Delivery Service), handling the communication problem of the IP, such as: duplication and data loss; network unavailability; delays; and unordered incoming packets. As drawback the protocol implementation is significantly more difficult.

The Reliability of the service is characterized by four main proprieties:

- Stream Orientation: when two applications transmit their data, the data flow to the destination sink comes in the same arrival order as it was generated by the source.
- Virtual Circuit Connection: from the developer and user viewpoint, the TCP service is like enabling a dedicated connection.
- Unstructed Stream: the TCP/IP stream service doesn't allow a structured data flow; it means that there's no way to distinguish the data flow records.
- Full-Duplex Connection: the connection generated by the TCP/service allows an independent and simultaneous data flux in both directions.

If the message to be transmitted is too big to be fit in a single TCP packet (usually 576 bytes taking into account the IP header), it is spitted in fixed length segment and then rebuilt once reached the destination, the fragment order is checked, and this operation is completely users transparent.

The reliability is achieved by the usage of acknowledgment messages (ACK) once a packet is received. The communication source holds a copy of each sent messages and remove them from the buffer just after the related ACK from the sink. In the simples and optimized configuration the source waits, once sent a packet, for its ACK before any further transmission; a packet timers is also set in the order to retransmit a message if a time-out occurs. This stop and wait approach is highly band inefficient, since there a band unused while waiting the ACK feedback.

2.3.1. TCP segment

The TCP segment consists of a segment header and a data section. Are now listed most valuable fields, Figure 3:

- Source Port: the 16-bit port number of the process that originated the TCP segment on the source device.
- Destination Port: the 16-bit port number of the process that is the ultimate intended recipient of the message on the destination device.
- Sequence number: For normal transmissions, the sequence number of the first byte of data in this segment. In a connection request (SYN) message, this carries the initial sequence number (ISN) of the source TCP. The first byte of data will be given the next sequence number after the contents of this field.
- Acknowledgment number (32 bits): When the ACK bit is set, this segment is serving as an acknowledgment (in addition to other possible duties) and this field contains the sequence number the source is next expecting the destination to send.
- Control bits (6 bits): URG (Urgent Bit) if triggered the priority data transfer is invoked on the segment; ACK if set one indicates that the segment is carrying an acknowledgment in the Acknowledgment Number field; PSH (Push Bit); RST (Reset Bit); SYN (Synchronize Bit) is a request to synchronize sequence

numbers and establish a connection relative to the Sequence Number field; FIN (Finish bit) to trigger the releasing of the connection.

- Windows size: Indicates the number of octets of data the sender of this segment is willing to accept from the receiver at one time. This normally corresponds to the current size of the buffer allocated to accept data for this connection.

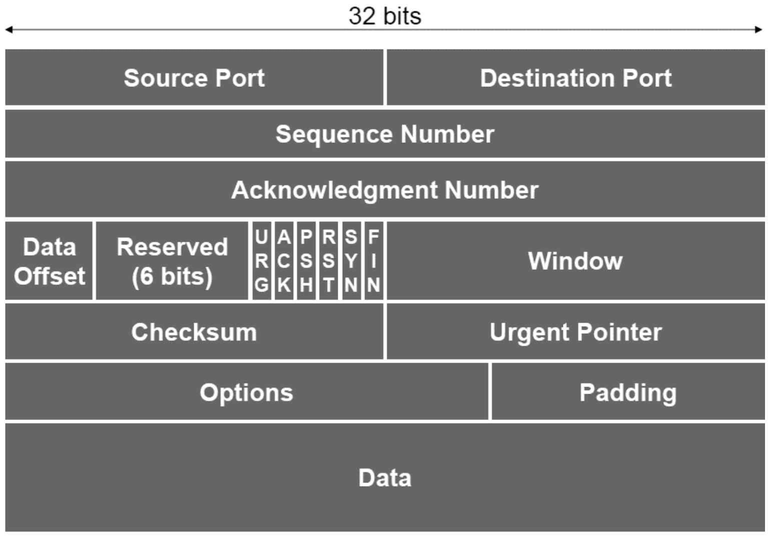

Figure 3: TCP Segment

2.3.2. Congestion control

Two are the main issues in the TCP protocol: the network congestion and the timeout/re-transmission mechanism. The congestion is a critical state of the network due to the overloading of datagrams inside one or more switching points (routers). When a congestions occurs, the packet serving delay grows and the datagrams start being inserted in queues, until they are not processed. In the worst case scenario the number of datagrams grows until the router's buffer maximum capacity and packets start to be lost.

From the hosts viewpoint, the congestion is a delay increasing. Moreover due to the lack of ACKs from the sent datagrams the sources will send again their datagrams exasperating the network congestion. The arise in the traffic creates a delay growing, which leads again to a Congestion Collapse scenario. Two methods to handle the network congestion problem: recover the network functionalities once the congestion happened (Recovery) or to avoid it (Avoidance). To avoid the network collapse, the TCP could utilize the Multiplicative Decrease Congestion Avoidance. The TCP/IP holds a second limit, beside the receiving window, exists the congestion window limit; in each instant the TCP assumes as source transmission window, the minimum between the two. Ideally the two windows have the same length, but in congestion scenario, the congestion window reduce, haling it's size, each time a datagram is lost (until the size equal to 1 limit). The transmission rate is then exponentially reduced and the timeout value is doubled at each loss. To recovery the normal network functionalities, once the collapse happened, the TCP can implement a Slow Start Recovery technique. In Slow Start Recovery once a new connection is established or the traffic grows again after a congestion event, the congestion windows has the size of a single segment and each time an ACK is received the congestion windows is raised of one. This mechanism allows to double the congestion window, each time all the ACKs are fetched back at the source, until the receiving windows limit. To avoid a too high increment rate, and having congestion again, the TCP protocol has another restriction: once reached the half of the receiving windows the increment rate is decreased; the congestion windows is augmented of one each time all the ACKs from the previous datagrams are gained. This combination of Recovery and Avoidance improves the TCP performances without adding of further congestion control tools.

3. Long Term Evolution Networks

Long Term Evolution (LTE), introduced in 3GPP Release 8, represents a significant change to the 3G UMTS/HSPA radio access and core networks. In this chapter will be presented the main LTE Architecture and characteristics.

3.1. Architectural Overview

In the LTE architecture the former circuit switch and packet switch system, already used in GSM and UMTS, is overcame; The Core Network is completely IP packet switching designed. In The LTE architecture is introduced a new radio access, L'E-UTRA (Evolved UMTS Radio Access Network), and a new Core Network, L'EPC (Evolved Packet Core) Figure. The radio access relays on a single element, the eNB (evolved NodeB) which has both roles of NodeB and RNC [7].

The eNodeB working with the Core Network (CN) and the adjacent eNbs is responsible of radio communication to and from terminal, the handovers and the session end procedures. Moreover in the handover procedure, which is automatically handled by the eNodeBs, the hierarchical higher layers are involved just at the end of the procedure for the notification of eNodeB change.

The eNodeB from now on acquire the security task form the previous RNC, due to that a new network elements, the Secure Gateway, is required in the order for allowing a secure communication, with the IPSEC protocol, between eNodeB and the Core Network. The Core Network EPC represents a perfect separation between the control and transport functions.

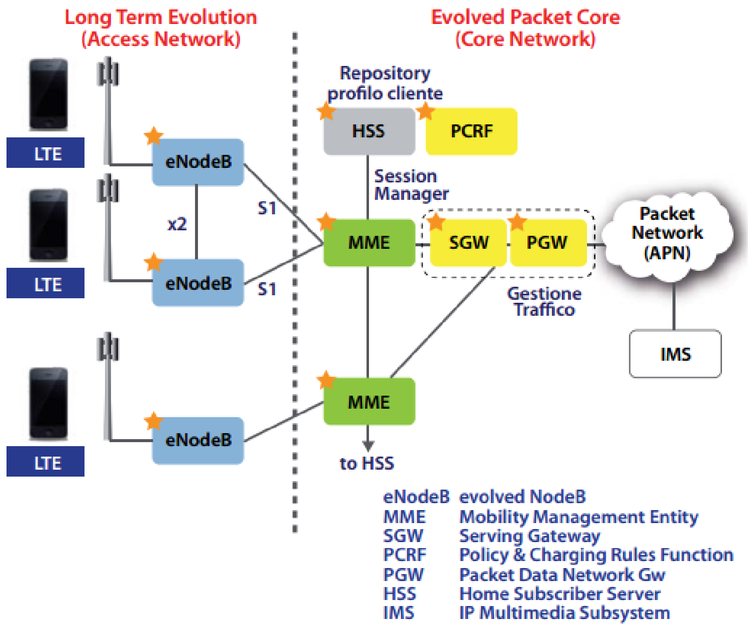

Figure 4: LTE architecture

The Evolved Packet Core is composed by the following descripted elements. The Mobility Management Entity (MME) is a control node, with functionalities similar to the SGSN: it's responsible for the terminals authentication procedures on the Home Subscriber Server (HSS), the network database, for the connections installation (Connection Management), their maintenance and their release and for the mobility handling (Mobility Management) for which the destination eNodeB receives the handover acknowledgment [8]. The suppression of the RNC requires the MME to handle the sending of the calling signal (paging) thorough the radio access.

The serving Gateway is a transport node with SGSN transport similar functionalities. It works with the PGW (PDN Gateway) for the user data transport handling the mobility of users between 3GPp accesses (e.g. E-UTRAN and 2G/3G). The PGW is a transport node with GGSN like functionalities, it is responsible to act as an "anchor" of mobility between 3GPP and non-3GPP technologies. PGW provides connectivity from the UE to external PDN by being the point of entry or exit of traffic for the UE. Moreover the PGW manages policy enforcement, packet filtration for users. The PGW assign to the UE an IP address in the user registration phase. Such IP address will be utilized from the network for all the user's procedures. Since in LTE all the services are packet centric the IP assignation is thought to be *always on* so the IP address is reserved for the users until the devices is not switched off. The PGW works with the PCRF to handle user's profiles. PCRF (Policy and Charging Rules Function) is responsible for the user and service policy (QoS, cargig, gating). To handle its task the PCRF uses the FCEF (Policy and Charging Enforcement Function) functionalities inside the PDN GW. Finally the HSS maintains the users' location information. In the following Figure 5 are portrayed and sorted the changing in the Networks Architectural from 3G UMTS/HSPA to LTE Architectures.

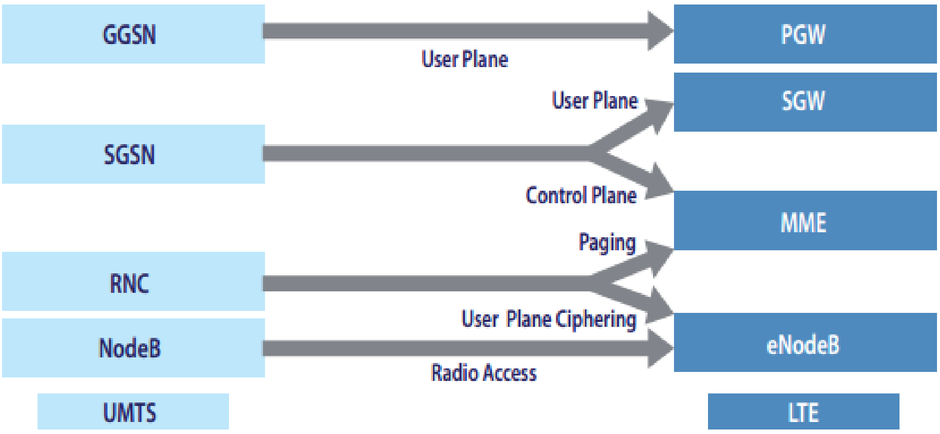

Figure 5 : Functionality migration from UMTS to LTE

3.2. Underlined technologies

The LTE networks comes with fulfilment duty of an large range of requirements, here are listed and studied the capabilities that are built-in in different categories of LTE devices in Releases 8 and 9.

3.2.1. LTE air interface

The LTE air interface uses a transmission time interval of 1 ms which, combined with features such as hybrid-ARQ, is designed to provide low latency. Each LTE frame (10ms) is dived in 10 sub frames or Time Transmission Interval (TTI) 1ms length which is also dived in two 0,5ms slots. Each slot is then divided in the frequency domain into a number of resource blocks, given by the channel bandwidth. A resource block contains 12 subcarriers from each OFDM symbol [9]. The number of OFDM symbols in a resource block depends on the cyclic prefix used. The resource block is the main unit used to schedule transmissions over the air interface Figure 6. The access to the RBs is performed through OFDMA in downlink and SC-FDMA for the uplink.

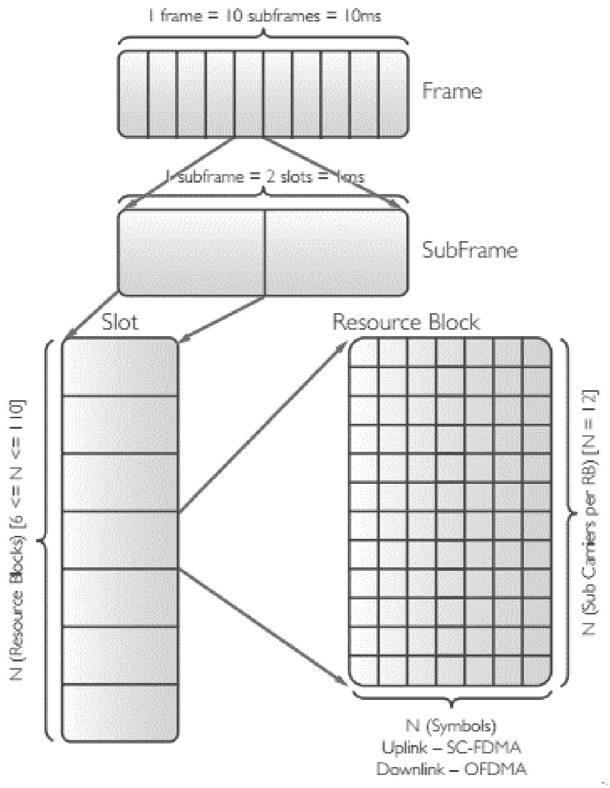

Figure 6: LTE Air Interface Elements

3.2.2. Radio protocol

The overall LTE radio interface protocol scheme is shown in, covering only the protocol part of the radio access in LTE. The physical layer carries the transport channels provided by the MAC layer. The MAC layer offers the logical channels to the Radio Link Control Layer (RLC). The logical channels define the type of data to be transmitted. On top of the RLC layer there is the Packet Data Convergence Protocol layer (PDCP), now used for both the control and the user plane, unlike in the WCDMA where it was user data plane oriented. Layer 2 provides upwards radio bearers, Signalling Radio Bearers (SRBs) carry the RRC signalling messages while the user plane Radio Bearers (RBs) carry the user data. Protocol architecture for the control plane, MAC, RLC, PDCP and RRC are all located in the eNodeB. In WCDMA architecture RRC and PDCP were both in the RNC, and the MAC layer was either in a NodeB (especially for HSPA) or in a Radio Network Controller (RNC) in case of Release'99 WCDMA [10].

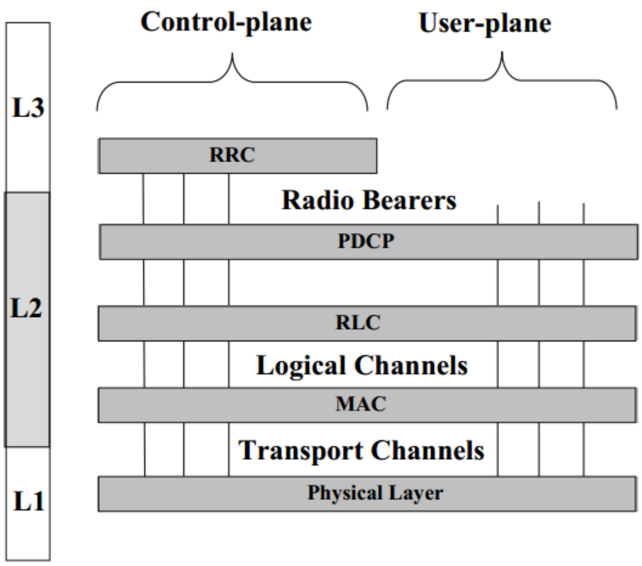

Figure 7: LTE Radio Protocol Stack

Medium Access Control

The MAC layer maps the logical channels to transport channels, as shown in Figure 8: MAC Layer. The MAC layer performs the following tasks:

- MAC layer multiplexing/de-multiplexing of RLC Payload Data Units (PDUs) belonging to one or different radio bearers into/from transport Blocks (TB) delivered to/from the physical layer on transport channels; also the Padding if a PDU is not fully filled with data [11].

- Traffic volume measurement reporting in the order to provide the RRC layer statistics about the traffic volume experienced.
- Error correction through Hybrid Automatic Repeat Request (HARQ), to control the uplink and downlink physical layer retransmission handling at the eNodeB.
- Priority handling between UEs by means of dynamic scheduling, thus the scheduling in the eNodeB is considered as MAC layer functionality.

The ciphering WCDMA functionality in the MAC layer are removed, neither is there transport channel type switching as the user data are only transmitted over a single type of transport channel (Uplink Shared Channel (*UL-SCH*) or Downlink Shared Channel (*DL-SCH*)). The user identification is based on the physical layer signalling, then there is no need to use the MAC layer for UE identification. The downlink MAC layer functionality in Figure 8 is specular in the uplink direction, but obviously the Broadcast Control Channel (BCCH) and Paging Control Channel (PCCH) are not present there and only one user is considered in the UE uplink MAC structure.

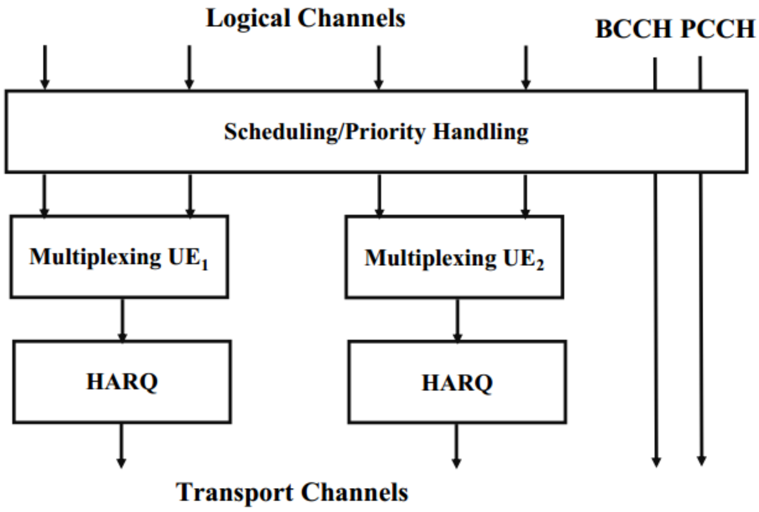

Figure 8: MAC Layer

The MAC layer provides the service to the RLC layer by logical channels. Different logical channels are defined for different data transfer services in the uplink and downlink directions.

Radio Link Control Layer

The RLC takes care of the following tasks:

- Gating the PDUs incoming from the higher layer;
- Error correction with ARQ, concatenation/segmentation, in-sequence delivery and duplicate detection may be applied, depending on the RLC mode used (described below);
- Protocol error handling.

The RLC layer can operate in three different modes:

1. *Transparent Mode* (TM): the RLC only takes care of delivering and receiving the PDUs on a logical channel but does not add any headers. The TM mode of operation is only suited for services that

do not use physical layer retransmissions or that are not sensitive to delivery order.

2. *Unacknowledged Mode* (UM): more functionality are provided such as in-sequence delivery of data, which might be received out of sequence due to HARQ operation in lower layers. The UM Data (UMD) are segmented or concatenated to suitable size RLC SDUs and the UMD header is then added. The RLC UM header includes the sequence number for in-sequence delivery (as well as duplicate recognitions).
3. *Acknowledged Mode* (AM): of RLC operation provides, in addition to the UM mode functionalities, also retransmission if PDUs are lost as a result of operations in the lower layers.

Packet Data Convergence Protocol

The Packet Data Convergence Protocol (PDCP) is located above the RLC layer. The key difference to WCDMA is that now all user data go via the PDCP layer, because ciphering is now in the PDCP, which is located in the eNodeB. The PDCP has the following functionalities:

- Header compression and corresponding decompression of the IP packets. This is based on the Robust Header Compression (ROHC) protocol. Header compression is more important for smaller IP packets in question since the large IP header could be a significant source of overhead for small data rates [12].
- Ciphering and deciphering user plane and control plane data. This functionality was located in the MAC and RLC layers in the previous WCDMA architecture.

3.2.3. Multicarrier Technology

The first LTE milestone was marked by the multi carrier approach selection in the network designing, both in *downlink* and *uplink*. The selected model were Orthogonal Frequency-Division Multiple Access (OFDMA) and Single-Carrier Frequency-Division Multiple Access (SC-FDMA) [13], in this section is illustrated those technology with a special focus on the uplink SC-FDMA that has a major importance in this report.

OFDMA

The frequency spectrum is arranged in subcarriers which are among them orthogonal (by them self or sorted in groups) to carry independent information streams. This approach leads to some key advantage lead to achievable in LTE networks. First of all it unleash a different bandwidth allocation to different users in the network in a fluid way, since no system parameters has to be modified. More over the orthogonal carrier approach ensure that the frequency re-use among different cell is facilitated and simplified.

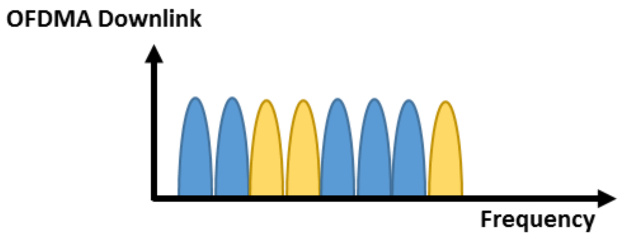

Figure 9: OFDMA LTE Downlink scheme

In LTE the sub-carrier spacing is 15 kHz regardless of the total transmission bandwidth. Different sub-carriers are orthogonal to each other, as at the sampling instant of a single subcarrier the other sub-carriers have a zero value, as was shown in Figure 10

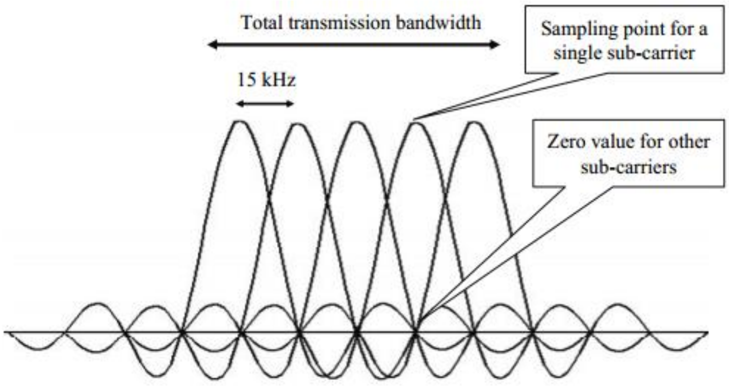

Figure 10: OFDMA sub-carriers' orthogonality

In Figure 11 the OFDM modulation scheme is presented: the data source bits feed the serial-to parallel conversion and then the IFFT block. Each input for the IFFT block corresponds to the input representing a particular sub-carrier and can be modulated independently of the other sub-carriers. The IFFT block is followed by adding the cyclic extension (cyclic prefix), this is due to avoid inter-symbol interference since adding a cyclic extension (longer than the channel impulse response) allows to avoid the effect of the previous symbol sent. The cyclic prefix is added by copying part of the symbol at the end and attaching it to the beginning of the symbol.

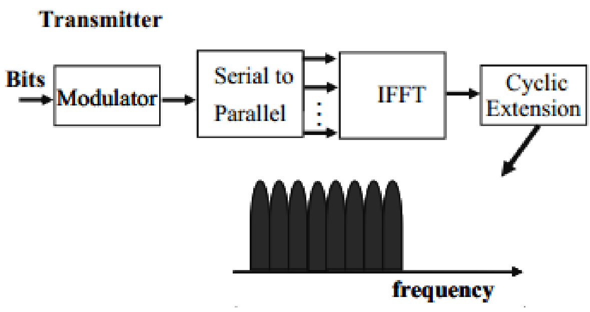

Figure 11: OFDMA Modulation Scheme

SC-FDMA

Single-Carrier Frequency-Division Multiple Access was chosen for the uplink set in LTE. This choice is due to the high Peak-to-Avarage Power ratio of the OFDMA approach, which is pointless to be achived on mobile devices that for their nature have to undercome energy efficency thresholds [14]. Moreover a new problem arise utilizing the OFDMA for the cellular uplink since in that senario the trasmission derives from offset in frequency references among the different simultaneosly tramisston deviecs. The orthogonality of the trasmission is seriosly compromized by the frequency offset and that lead to multiple access interferece.

As in OFDMA, the transmitters in an SC-FDMA system use different orthogonal frequencies (subcarriers) to transmit information symbols. However, they transmit the subcarriers sequentially, rather than in parallel, as shown in Figure 12, which leads to a reduction of the envelop fluctuation in the waveform.

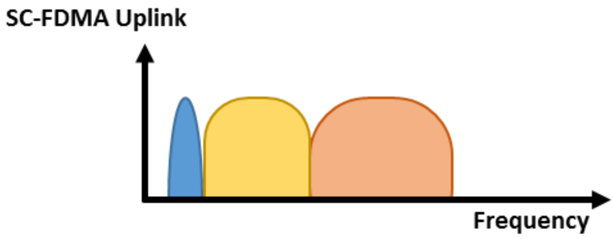

Figure 12 : SC-FDMA LTE uplink scheme

The coding scheme used in the SC-FDMA are binary phase shift keying (BPSK), quaternary PSK (QPSK), 16 level quadrature amplitude modulation (16-QAM) and 64-QAM which are selected by the device based on the channel state. After that at the transmitter, shown in Figure 13, side the x_n symbols are grouped in N-symbols blocks.

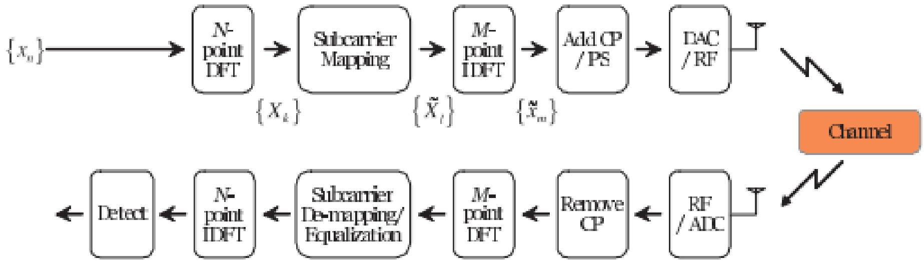

Figure 13 : Transmitter and receiver in SC-FDMA

The following step is to modulate the SC-FDMA subcarriers into perform an N-point discrete Fourier Transform (DFT) in the order to produce a frequency domain representation x_k of the input symbols. Then each one of the N DFT output is mapped to one of the M (M>N) subcarriers. As in OFDMA, an M-point inverse DFT (IDFT) transforms the subcarrier amplitudes to a complex time domain signal $\widetilde{X_m}$. Each $\widetilde{X_m}$ then modulate a single frequency carrier and all the modulated symbols are transmitted sequentially. The following block in the chain inserts a set of symbols referred to as a cyclic prefix (CP) in order to provide a guard time to prevent inter-symbol interference (ISI) due to multipath propagation. The cyclic prefix duration is $5.2\mu s$ (for the first symbol in each slot) and $4.9\mu s$ (for all the other symbols) [11] their use transforms the liner convolution of the multipath channel into a circular convolution, enabling the receiver to equalize the channel by scaling each subcarrier by a complex gain factor.

Then a linear filtering operation, referred to as pulse shaping in order to reduce out-of-band signal energy, is performed.

4. NS-3 network simulator

Ns-3 is a discreet event network simulator, designed built from an international community for research goals, it's carried and minded to a open-source approach [2]. That's approach become necessary since the nowadays network complexity raises with a ratio which doesn't allow any other traditional from the top theoretical study. O Ns-3 key feature is its modularity, which allows the integration, the adding and the shaping of each components used in the different simulation scenarios. More over in this elaborate the focus is mainly on the Ns-3 LTE module, called LTE-EPC (LENA), which is developed and maintained by the *Centre Tecnològic de Telecomunicacions de Catalunya* (CTTC) [15].

All the simulations and results were made with the 3.24 Ns-3 simulator version and the v8 release of the LENA module.

4.1. The LENA Module

An overview of the LENA module is shown in Figure 14, where can be seen two different main components: the first one (LTE Module, pink link) includes the protocol stack (RRC, PDCP, RLC, MAC, PHY) and all the communication between UEs and eNodeB station methods; the second

(EPC Model, blue lines and green lines) contains the communication the network interface and the intercell coordination.

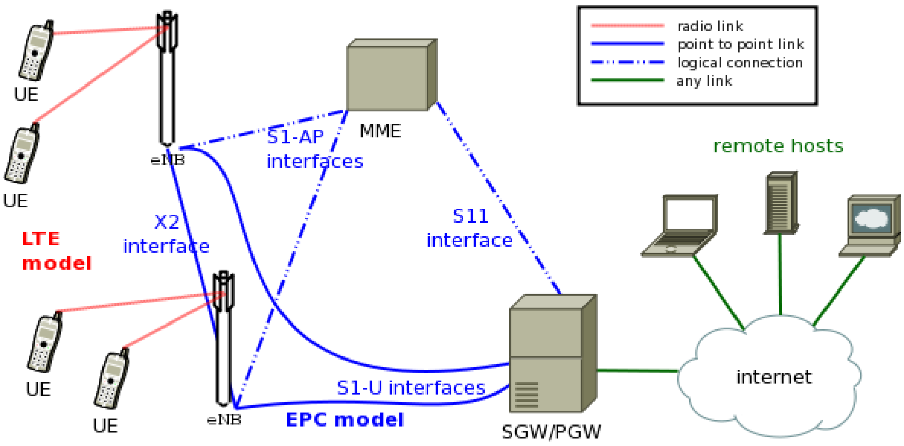

Figure 14 Overview of the LTE-EPC simulation model

4.1.1. Spectrum management

The main carrier

The main carrier in downlink is marked in the simulator by fc and its stands for the EUTRA Absolute Radio Frequency Channel Number (EARFCN), a number in the [0, 65535] interval [16]. A bilateral link holds between the EARFNC and the main carrier in MHz, it's given by the following relations:

$$F_{DL} = F_{DL-low} + 0.1(N_{DL} - N_{OFFS-DL})$$

$$N_{DL} = 0.1(F_{DL} - F_{DL-low}) + N_{OFFS-DL}$$

Where F_{DL} stands for the main carrier in MHz, N_{DL} stands for the Downlink EARFCN (*E-UTRA Absolute Radio Frequency Channel Number*) and then $N_{OFFS-DL}$ and F_{DL-low} are table values [16]. Following the Table 5.7.3-1 [16] the parameters are the followings:

- $N_{OFFS-DL} = 0$
- F_{DL-low} = 2110

$$F_{DL} = 2110 + 0.1(N_{DL})$$
$$N_{DL} = 10 \cdot (F_{DL} - 2110)$$

In this paper the main focus will be on the *uplink* implementation, rather than the *downlink*, then the main carrier can be find by a shifting operation

$$N_{UL} = N_{DL} + 18000$$

$$F_{UL} = F_{UL-low} + 0.1(N_{UL} - N_{OFFS-UL})$$

Bandwidth

To follow the granularity requirements, due to the minimal resource allocation manageable is an RB [17], the bandwidth is shown by RBs and its can have, due to the standard [18] the 6, 12, 25, 50, 75, 100 values. For the simulation carried on through this report the RB value is set to 25.

Each duple $(fc\ , B)$ defines the spectrum model that could be assigned to an eNodeB. Speaking of orthogonal management the cells don’t generate any intercell interference, then the SINR calculus of each RB is performed taking in account the useful power and the channel noise. Instead management isn’t set to orthogonal part of the bandwidth will overlap. The SINR calculus, in Ns-3, is performed on each RB each TTI so the overlapped RBs will have a smaller SINR than the not overlapped ones. In this report the scenario takes into account just one eNodeB scenario so no problem of overlapping are taken into account.

Ns-3 allows to define a mobility model for the simulated nodes but in the report scenario the nodes are supposed to be still.

4.1.2. Scheduler

The scheduler, which as a main role in this report, is an algorithm which handles the division of the available resources among the users in the system according to a certain criteria. Moreover, is the available bandwidth is composed by *K* RBs, and N users are active, the scheduler has the goal to assign the *K* RBs to the N users. Different schedulers are implemented in NS-3 such us Round Robin (RR), Proportional Fair Scheduler (PF) and Maximal Throughput Scheduler (MTS) [19]. All the schedulers will be further analysed and discussed in the in Chapter 5.

4.1.3. Simulations

A simulation of a LTE scenario follows the following steps and characteristics:

- *Devices parameters*: consists in defining the physical coordinates of the base stations (eNodeB in the LTE scenario) and UE terminals. More over transmission powers and eNodeB characteristics are defined.
- *Channel settings*: the radio channel is modelled to fit the study bias. Fading traces can be added to the simulation, vehicular/pedestrian/urban fading traces are already built inside Ns-3 or they could be generated by a Matlab file also included.
- *Internet and Ipv4 Helpers*: the internet protocol stack has to be installed on the nodes which have to be granted an IPv4 address.
- *Simulation and results collections*: the simulation program is written using the C++ language under the Eclipse IDE, which is

previously set to work on the Ns-3 framework. The collected results are listed and shown in following paragraph.

4.1.4. Results collections

Once a simulation ends Ns-3 generates a different kind of output files, such as:

- *RlcStats.txt*, both in *uplink* and *downlink*, is used to evaluate the transmitted bytes inside the simulation time interval and their delay;

$$Throughput = \frac{8\ \cdot TxBytes}{SimTime\ \cdot 10^6}$$

- *MacStats.txt*, both in *uplink* and *downlink*, is used to evaluate different aspects like the Modulation Coding Scheme of each block, which lead to study the spectral efficiency and the channel quality.
- *Transport.pcap* files, could be triggered in the simulation to trace all the packet at the transport layer on a particular device. The results are be analysed thought different software such as Wireshark, Tcpdumb, TcpAnalyzer and Matlab [20] .

Moreover our focus will be on the TCP traffic analysis since nowadays the LTE human traffic is carried by html, which relays on TCP protocol. In the order to achieve that from the *Transport.pcap* are collected for each packets the following data:

- *frame.number*: which gives the id of the captured frame;
- *frame.time:* gives the time stamp of the captured packet from the 1st January 1970 (as to be parsed in the order to be used in Matlab for further investigations);
- *frame.len*: the lengths of the captured frame;
- *ip.src*: the communication source;

- *ip.dst*: the communication sink;
- *tcp.seq*: the sequence number in the TCP header of the captured packet;
- *tcp.acknum and tcp.ack:* 32 bits which picture the Acknowledgment Number. This number ensure the reception of the previous segments and says to the communication source the following sequence packet which is expected to reach the sink. The Acknowledgment Number filed has meaning if and only if the TCP Flag is set to 1.
- *tcp.analysis.akt_rtt:* To write it if we will use the round trip time

5. LTE Schedulers

The limited capacity of the second-generation cellular systems has marked the forced path that industries and standardization entities have to go thoughts. The 3GPP LTE offers higher capacity and more flexible radio resource management along with high speed packet access data technologies. However, LTE has been designed for a broadband scenario, while most M2M applications transmit and receive small amounts of data leading to an unreasonable ratio between payload and required control information and monopolized transmission protocols. Is now shown how the *uplink* scheduler framework behave in LTE.

5.1. Signalling and Scheduling

In 3GPP LTE the scheduling takes place at the base station side (eNodeB) and the allocation decision are sent to the UEs through appropriate control channels. In particular each UEs sends scheduling request to the eNodeB thought L1/L2 control signalling asking for access to the UL shared channel [21]. Each UE is assigned one physical uplink control channel (PUCCH), therefore, at the presence of a large number of machines, as in M2M scenarios, a small numbers of PUCCH resources is possible. As the number of devices grows, so does the associated signalling load. Based on the scheduling requests, the eNB decides the PRB-to-UEs (or M2Ms) allocation at each TTI and sends this information to the M2Ms through the corresponding physical DL control channel (PDCCH). The PDCCH physical channel is loaded into the first one to three orthogonal frequency division multiplexing (OFDM) symbols of the DL time–frequency grid, consuming system resources. According to the 3GPP specifications, up to ten UEs (or MTCDs) may be loaded in a single subframe. Hence, PDCCH is unable to support hundreds of MTCDs demanding simultaneous access to the shared channel in future M2M scenarios. It is obvious from the above discussion that the signalling for supporting UL scheduling in M2M scenarios becomes prohibitive, and the current mechanisms may not effectively support MTCDs in LTE cells. Modifications on existing approaches as well as the design of new solutions for reducing the signalling overhead are important in future M2M-enabled LTE systems.

In general, scheduling is not part of the standardization work, rather it is an implementation-specific issue. That's why become each network carrier should use the scheduler which fits to its usage needs. However, signalling is standardized, thus any scheduling proposal should be in line with the set of control requirements. Toward this purpose, several schemes have been devised for dynamically allocating resources to UEs with heterogeneous QoS requirements. A set of nine QoS classes has been prescribed in 3GPP specifications [QoS class identifiers (QCIs)], classifying services (or radio bearers) based on the resource type [guaranteed bit rate (GBR)/non-GBR], priority order, packet delay budget, and packet loss rate characteristics. Due to the new M2M scenarios coming into play the scheduling entities have to deal with extremely diverse QoS criteria. For example, delay tolerance may span from tens of milliseconds, for real time related machines, to several minutes. Thus, forming specific QoS classes/clusters is not an easy task. As scheduling exploits channel and traffic dynamics, the existence of a big number of M2Ms may induce further processing delays and impose storage constraints into the base station. Developing practical scheduling schemes that support a large number of MTCDs without deteriorating the performance of standard LTE services is therefore a challenging issue.

In the following paragraphs are presented the literature *uplink* schedulers. In the Ns-3 Simulator all the *uplink* scheduler are, nowadays, modelled as Round Robbin, while their *downlink* ones are not. That's fact is due to design choices. This work thesis work is then addressed to feel the gap in the NS-3 implementation pattern thought the developing of the missing *uplink* schedulers.

Round Robin Scheduler

The Round Robin (RR) scheduler is the oldest and simplest (keeping a good scheduling fairness) scheduler found in the literature. It works by dividing the available resources among the active users flow waiting with a non-empty queue. If the number of RBs is greater than the number of active flows, all the flows can be allocated in the same subframe. Otherwise not all the flows can be scheduled and they are served to in a First in First out way. Then, in the next subframe the allocation will start from the last flow that was not allocated.

Maximal Throughput Scheduler

The Maximum Throughput Scheduler (MTS) schedules the user with most favourable channel conditions in the TTI of index k [22]. Its scheduling metric is given as

$$i_{MTS}(k) = \arg \max_{1 \le i \le N} r_i(k)$$

Where $r_i(k)$the instantaneous is rate of user *i*. The MTS can achieve maximum cell throughput, but for extremely unbalanced average SINR distributions of users, it could result in the starvation of UEs experiencing bad channel conditions.

BETS Scheduler

Blind Equal Throughput Scheduler is channel-unaware scheduler which guarantees equal throughput for all users by scheduling the users according to the metric [23]

$$i_{BETS}(k) = \arg \max_{1 \le i \le N} \frac{1}{\varsigma_i(k)}$$

where $\varsigma_i(k)$ is the past average throughput of the *i-th* UE at TTI k, which is given by (see [11])

$$\varsigma_i(k) = \beta \cdot \varsigma_i(k-1) + (1-\beta) \cdot r_i(k)$$

where $\beta \in [0, 1]$. However due to its channel-unaware approach is not very efficient in terms of cell throughput.

PFS Scheduler

The Proportional Fair Scheduler (PFS) can improve the cell throughput by incorporating channel conditions, of the MTS, in the BETS in the following way: [11]

$$i_{PFS}(k) = \arg \max_{1 \le i \le N} \frac{r_i(k)}{\varsigma_i(k)}$$

Scheduling Algorithm #1 for M2M Communications

This *uplink* scheduler is designed to address the problems raised by the M2M communication pattern [24]. It is here explained the RBs allocation algorithm behaviour:

- Step 1: considering the set of available RBs in the current transmission time interval.
- Step 2: assigning the RBs prioritizing the LTE users.
- Step 3: if there are no RBs, after step 2, end the allocation and start again from step1. Otherwise sort the M2M devices according to their achievable ratio $r_i(k)$.
- Step 4: Allocate each RBs of the set to the best device only if its maximum delay tolerance is smaller than the mean delay tolerance of all the remaining MTC devices asking for access the channel divided by 2, reduce by one the F_k, which is the number of RBs requested by the k device. If the selected M2M device doesn't comply with the specific it has to be insert in set of inactive devices for a number of TTI equal to its waiting time divided by 2.

- Step 5: if F_k becomes equal to zero the selected M2M is excluded from further allocations otherwise is checked its $r_i(k)$ allows to be assigned to the following *i-th* RB. Step 5 is performed again
- Step 6: Repeat step 3 for the left RBs. Then if there are still any RBs left, those can be assigned without any delay related concerning.

This algorithms promise to be really powerful and scalable for the future but is not drawbacks free. First of all is not shown by which politics the LTE users should be handled, in this report implementation will be supposed to use a Proportional Fair approach in the order to have a good tread off between fairness and user throughput. Another main drawback, and probably the most hard to be addressed, is how the eNodeBs, and the carriers, should be aware about the MTC delays. At the eNodeB side few data are available, such us users' transmission queue and SINR, but there isn't any way to store the delays for each device. Achieving that underling achieving effort at carrier's side to create such an architecture to store each device time needs, it could be a to big challenge due to the fact that beside a new structure to store those data the network will experience an overhead of signalling messages to handle that.

In the following Chapter is shown a new M2M scheduling algorithm that leverages over a self-network-awareness to identity the connected devices, both LTEs and M2Ms, for its scheduling decisions.

6. M2M Aware Scheduler

From network simulations arise the need of an *uplink* scheduler which should address the following proprieties:

- Device awareness;
- QoS minded;
- Resource awareness;
- Traffic awareness.

By device awareness is intended the ability of discriminate and group the connected devices in sets without any direct signalling messages exchange between the device and the eNodeB. This priority is of first importance when the number of M2M devices grows to the forecasted limits [1] in the order to avoid network congestions. The M2M Aware Scheduler groups the users in two main sets: LTE devices and MTC devices. Due to heterogeneous scenario, of which the MTC devices belong to, would be interesting group MTCs in more group according to their different needs in matter of delay sensitivity or message priority. QoS minded means being able to assign resources to human users before scheduling M2M ones. That's another milestone for network operators since human users are the first to be affected by the machines traffic, a way to avoid human's channel congestion has to be studied. Most of the schedulers present in literature lack on resource awareness. Usually the channel is granted to machines which comply with certain specifications (such as best achievable rate, or longer inter scheduling time) but no control is performed on their effective channel needs. Finally another aspect that is not taken into account is the traffic type. Nowadays, de facto most of the H2H traffic flows through TCP protocol (Http surfing) and as well the M2M traffic seams fit this type of traffic too, due to the TCP message reliability.

The M2M Aware Scheduler here presented tries to address those four proprieties.

6.1. Device awareness

At base station side just scheduling information such as transmitter queue, waiting time, SINR and packet arrival ratio are available, that force the network operator to guess the identity of each device trying to connect. The guessing mechanism is central for the scheduling procedure and leverages on knowing the mean number k of human users present in the cell. This knowledge could be easily get from study on the actual UMTS/GSM usage statistics. The first step is starting from the assumption, legitimated by literature reviews, that: M2M traffic is mainly *uplink* centric. The guessing algorithm steps are the following, performed at each TTI:

- Step 1: compute the ratio, for each user, between total traffic displaced in downlink and the one in uplink.
- Step 2: sort the users in descending order.
- Step 3: assuming the first k users to be human users.
- Step 4: schedule the users and update the statistics for the next TTI.

This sorting algorithm allows the eNodeB to know the nature each attached device in a relatively short interval as will be shown in the next chapter.

Moreover, and for future study (which are not traded in this report), different requirement sets (machine message priories) of M2M could be discovered if, after the third step, is added the following:

- Splitting the M2M in sets according to their packet arrival rate.

Once kwon that could be assumed that device with a high packet arrival rate have to be scheduler more frequently than the others.

Traffic awareness

6.2. Resource awareness

The resource utilization is central aim for a network operator and realize an uplink scheduler which is able to assign resources in such a way to maximize the channel usage is a priority. In the M2M Aware Scheduler this resource optimal assignment is performed with the use of the following described recursive algorithm.

Knowing k, the number of RBs per flow, and q, the users' transmission queue size, the number of bytes b that can fit the channel with k RBs is studied. The result is saved in a hash map with pair values k and b. $k = k - 1$ and the evaluation is performed again, for all the values k, until $k = 0$. After that the number of RBs to be assigned the user are taken from the hash map as the minimum number of k able to send b bytes in the queue. This mechanism, as will be shown in the Performance analysis chapter allows an increase in the channel usage.

6.3. Traffic awareness

Leveraging on the TCP traffic type the uplink scheduler is able to recognise the TCP ACKs from the LTE users, thanks to the device awareness algorithm, and prioritize them to be scheduled. That's allows LTE users to build their transmitting queue faster. Also the *downlink* is modified for two main reason:

- Allows M2M ACKs to be easier scheduled;
- Avoid wasting of channel bandwidth in over assignment of resources to M2M users.

The last point should be more analysed in future works due to the fact that inside the same TTI, in downlink schedulers, there is no awareness on how the previous resources are assigned, leasing to an over assignment of resources to users.

6.4. M2M Aware algorithm

Is here presented and explained the *uplink* algorithm behaviour scheme, Figure 15:

- Step 1: select the number of uplink flows, as the number of users with non-empty transmission queue willing to access the channel.
- Step 2: finding the available RBs per flow as $= \frac{RB_{num}}{active\ users}$. If $f <$ 3 the value of f is set to 3, this is to give a reasonable achievable rate each TTI.
- Step 3: Split the device in two groups utilizing the device aware algorithm.
- Step 4: Sort each group using the Proportionally Fair metric.
- Step 5: Serve the LTE user with the higher
$$i_{PFS}(k) = \arg\max_{1 \le i \le N} \frac{r_i(k)}{\varsigma_i(k)}$$
Assigning to it the minimum number of RBs *x* needed.
- Step 6: remove the actual user from the next allocation. If $x < f$ a number f - x are added to the free set of RBs and can be allocated for other devices.
- Step 7: continue from Step 5 until there aren't LTE users ready to transmit.
- Step 8: if *r*, the number of RBs still not allocated, is greater than zero the allocation of resources to the M2M devices can start.
- Step 9: Serve the M2M user with the higher
$$i_{PFS}(k) = \arg\max_{1 \le i \le N} \frac{r_i(k)}{\varsigma_i(k)}$$
Assigning to it the minimum number of RBs *x* needed.
- Step 10: remove the actual user from the next allocation. If $x < f$ a number f - x are added to the free set of RBs and can be allocated for other devices.
- Step 11: continue from Step 9 until there aren't M2M users ready to transmit or the number *r* of available resource is zero.

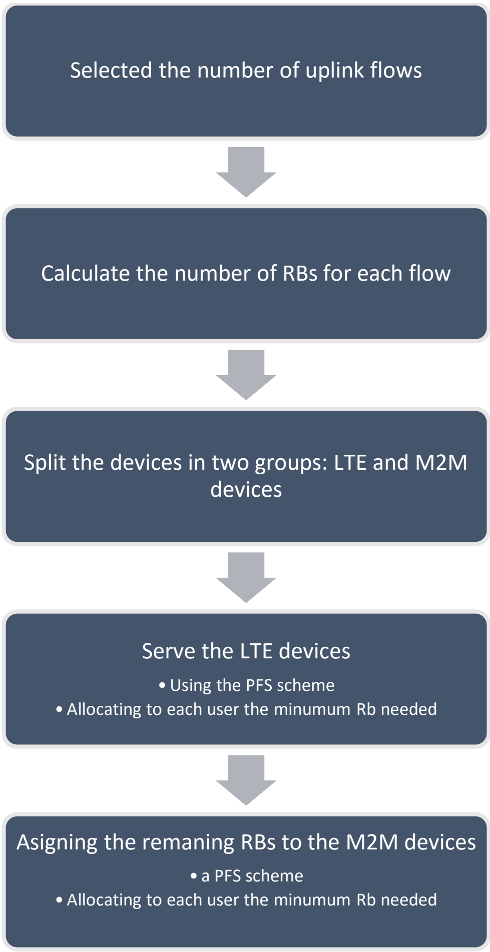

Figure 15: M2M Aware Scheduler Scheme

In the following chapter are analysed the performance of the M2M Aware Scheduler along with the literature ones.

7. Performance metrics and simulation results

In this chapter will be listed the performance metrics for M2M scenario uplink schedulers and then the simulation results are discussed.

7.1. Performance metrics

7.1.1. Average cell throughput

The average throughput is extracted by the Ns-3 simulator log files and calculated as follows:

$$Throughput = \frac{8 \cdot TxBytes}{SimTime \cdot 10^6}$$

First of all is studied the uplink throughput for the literature schedulers, without any M2M device, to prove the right implementation of those. After that M2M devices will be added to the scenario and will be studied the effect of those on the LTE human devices downlink throughput.

7.1.2. Jain's fairness index

To study and rate the fairness of each scheduler taken into account is used the Jain's fairness index J, which is first introduced in [25] as:

$$J = \frac{(\sum_{i=1}^{N} x_i)^2}{N \sum_{i=1}^{N} {x_i}^2}$$

Where x_i is the throughput achieved by the *i-th* UE. Note that J = 1 indicates perfect fairness among the users in the system. This metric is applied just to the initial study for the just LTE user scenario due to the fact that it doesn't hold anymore when taking into account devices with different QoS priorities.

7.1.3. Resource Blocks Utilization

The resource block utilization is a major goal for a scheduler and logging, in the NS-3 simulator, the utilization, each TTI instant, the RBU is studied as the ratio:

$$RBU_i = \frac{Trasmitter\ queue_i}{Maximum\ data\ tramittable}$$

Where $Trasmitter\ queue_i$ represents the queue of the *i-th* user. The study is carried both for the *uplink* and *downlink* and separated between LTEs and M2M resource block utilization.

7.1.4. Round trip time

Our test scenario is using the TCP protocol to perform the simulations. The TCP protocol relays on the reception of an ACK message after the data reception, at the destination side. This protocol require the knowledge of the RTT in the order to compute the maximum time for the data retransmission [26]. Usually the RTT is calculated averaging the time needed for the initial 3-handshake but since the TCP LTE sources are set to saturate the channel with their data there will be just the initial 3-handhake. Although the RTT will be calculated through the Wireshark software [27], according to:

$$RTT_{estimated} = (1-\propto)RTT_{estimated} + \propto RTT_{computed}$$

Where $\propto$ is constant.

7.2. Simulation setup

7.2.1. Uplink schedulers verification

This Ns-3 scenario is characterized by 10 LTE users flooding the channel and saturated it. In the order to perform the literature scheduler comparison. The channel is not affected from fading and underling to the Friis attenuation rule.

7.2.2. M2M dense scenario

Here fading is added to the scenario and along to the 10 LTE users are added 1, 5, 10, 25, 50, 100 M2M users and are performed measures to evaluate the LTEs channel deterioration. It’s important to under light that the forecasts [1] suggest that a dense M2M scenario will be composed by thousands of machine communication devices, with a really small packet arrival rate, but for the sake simulation less devices with and higher, 1 Kbyte/s , traffic rate are taken into account. From the system view point those M2M devices could be seen as aggregator points for the thousands of sensors sparse over the simulation area.

The M2M devices are simulated through a custom adaptation of the Ns-3 Bulk Sender which allows to schedule packets according to a Poisson traffic generation with exponentially distributed inter arrival time. The rate of packet generation is fixed to 10 Packets/s, each one with a 100 byte payload. From the literature is stated as the M2M traffic is centred over the point that each device sends a small amount of data with an generation time of minutes or hours, in this report is studied an generation rate of 10 packets/sec in the order to underline the effect of those devise over the human-to-human flows. From a practical view point each device generated could be seen as a M2M traffic aggregator which collects data from sensors in the area [28]. The distribution of the inter arrival time is shown in the following Figure 16.

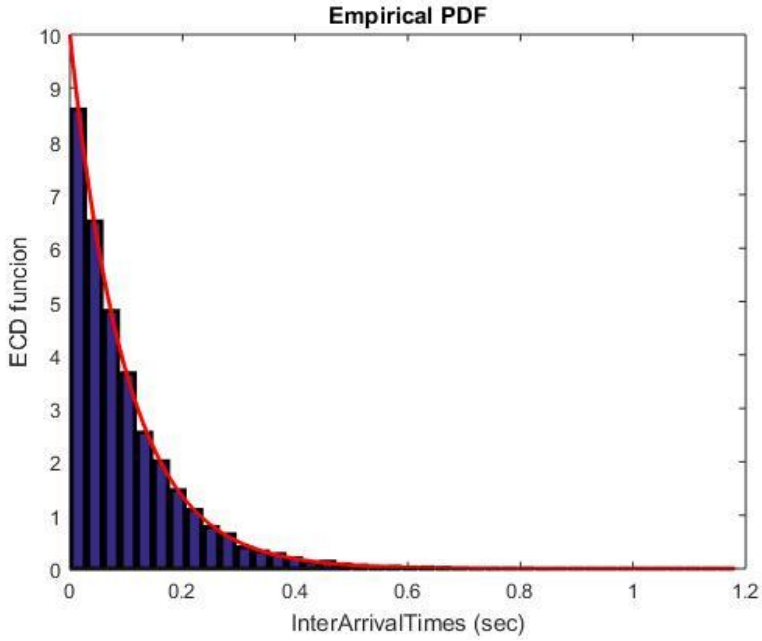

Figure 16: Exponentially distributed inter arrival time for a M2M device (10 packets/s)

The fading trace utilized is the vehicular scenario (speed of 60 km/h), embedded in the Ns-3 simulator, which simulates a Rayleigh fast fading traces with $v_d = 120$ Hz.

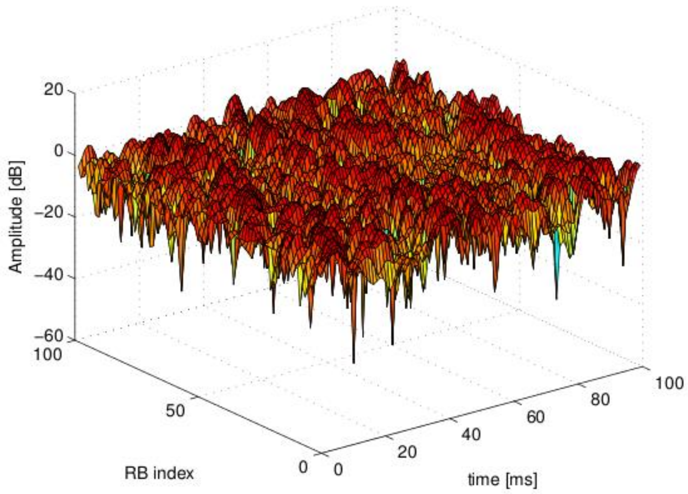

Figure 17:Excerpt of the fading trace included in the simulator for a vehicular scenario (speed of 60 km/h)

The system parameter are here listed below.

Parameter	Value
Number of RBs	25
Bandwidth	5 MHz
AMC Model	Piro
Error mode control	Deactivated
Radio Link Control Mode	Unacknowledged
Tx power of eNode	30 dBm
Tx power of UEs	23 dBm
Noise figure at eNode and UEs	5dB
M2M TCP packet size	100 Bytes
M2M packer geration rate	10 packet/s
Simulation duration	30 seconds (if not otherwise stated)

Table 2: System parameters

The device are located in cell of 1 km radius with the following specifications:

LTE	SINR (dB)	Distance (m)
1	132.47	999.99
2	63.23	967.40
3	57.46	936.62
4	54.17	907.5
5	51.90	879.88
6	50.19	853.65
7	48.82	828.68
8	47.69	804.89
9	46.73	782.18
10	45.91	760.48

Table 3: Human user's location

7.3. Simulation results

The Ns-3 simulation results are here shown and discussed.

7.3.1. Uplink schedulers verification

In the order to ensure the results reliability, all the *uplink* schedulers are implemented using as *downlink* the PFS. Here are tested with a 60 seconds network simulations and outcome are shown through Average cell throughput evaluation and Jain's fairness index. From the simulation results, Figure 18 and Figure 19, referred to Table 4 , the *uplink* schedulers are rightly implemented.

	MTS	BETS	PFS	RR
Cell throughput (Mbits/s)	4.927	3.485	3.807	3.428
Jain's fairness index	0.1	1	0.937	0.925

Table 4: Uplink schedulers comparison

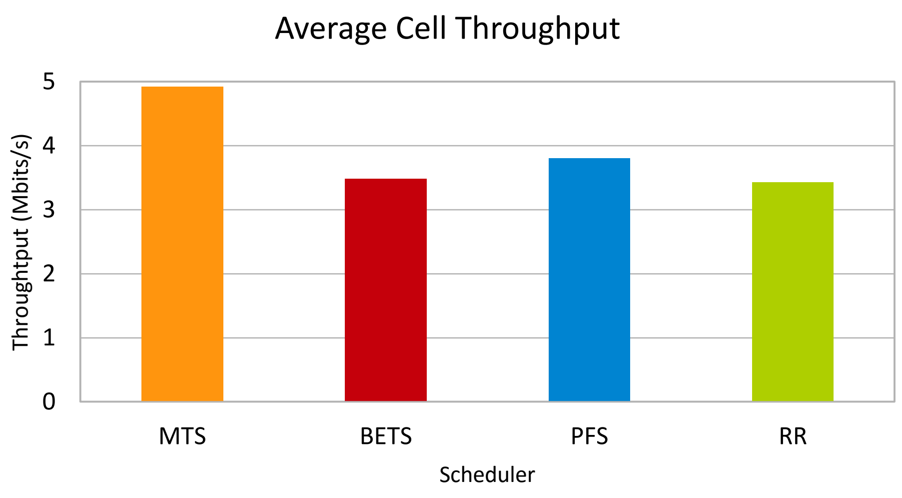

Figure 18: Average cell throughput with different uplink schedulers

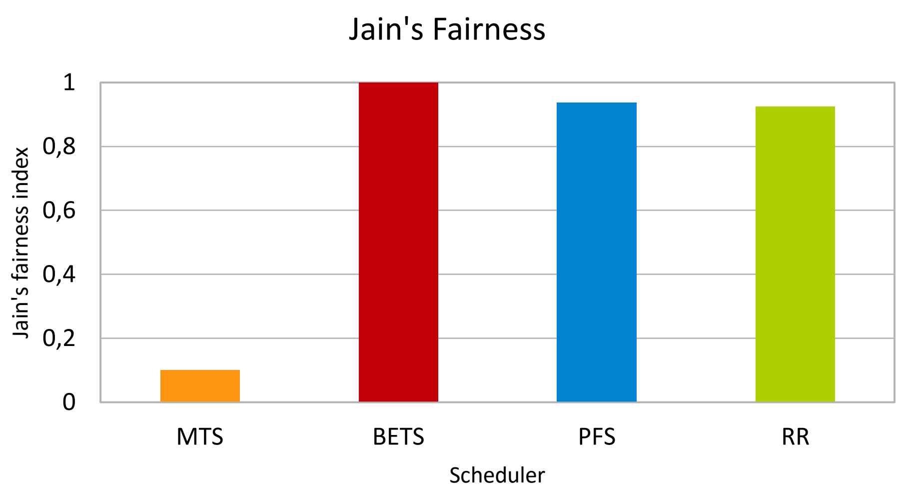

Figure 19: Jain's Fairness Index with different uplink schedulers

7.3.2. M2M dense scenario

Here are evaluated the simulation results for the M2M scenario.

Average cell throughput

First of all a simulations utilizing the classic NS-3 PFS scheduler (with *Round Robin* uplink module) are carried on and the LTE user channel results are shown in the Table 5. Each simulation is ran five times and its results are then averaged, due to the randomness taken from the M2M Sender Source module.

M2M	LTE users cell Throughput (Mbytes/s)	M2M message exchange (Kbytes)
0	15.495	0
1	15.403	30
5	15.071	142.3
10	13.568	289.1
25	11.242	732.7
50	11.204	1468.9
75	9.218	2200.9
100	7.223	2928.4

Table 5: Average Cell Throughput with M2M devices. PFS as selected scheduler

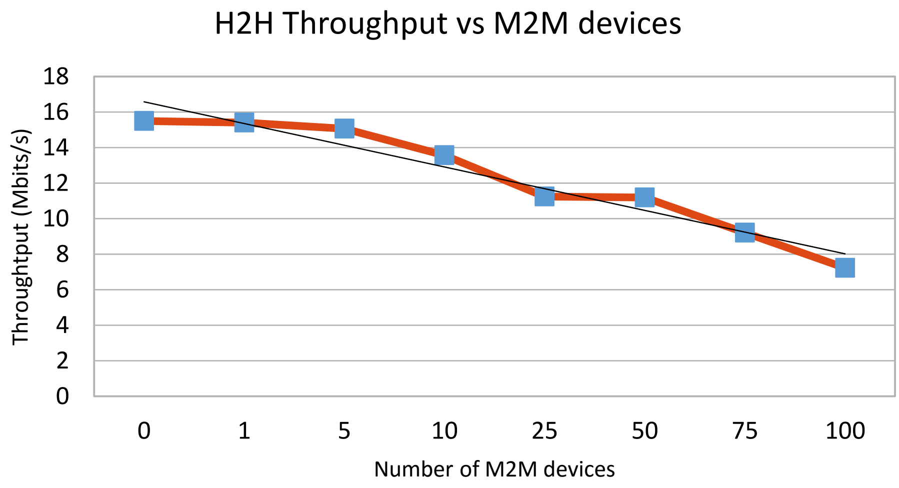

Figure 20: Average cell Throughput with M2M devices

The deterioration of the LTE throughput is high lined from the simulations and marks the first order importance of M2M related studies over LTE. From Figure 20 is easy to see how deep the M2M deterioration factor is:

$$LTE_{\ thoruhgput} = \alpha \cdot number_{M2M} + \beta \ \ (Mbyts/s)$$

With the two constants $\alpha = -1{,}2233$ and $\beta = 17{,}808$. This cell throughput behaviour is the starting point of this report works. Different uplink scheduler are now studied in the order to address to the problem and shape the LTE cell throughput behaviour under M2M scenario.

The following simulations set shows how the literature, the algorithm #1 for M2M [24] and the presented M2M Aware schedulers behaves in the 100 M2M scenario, Figure 21.

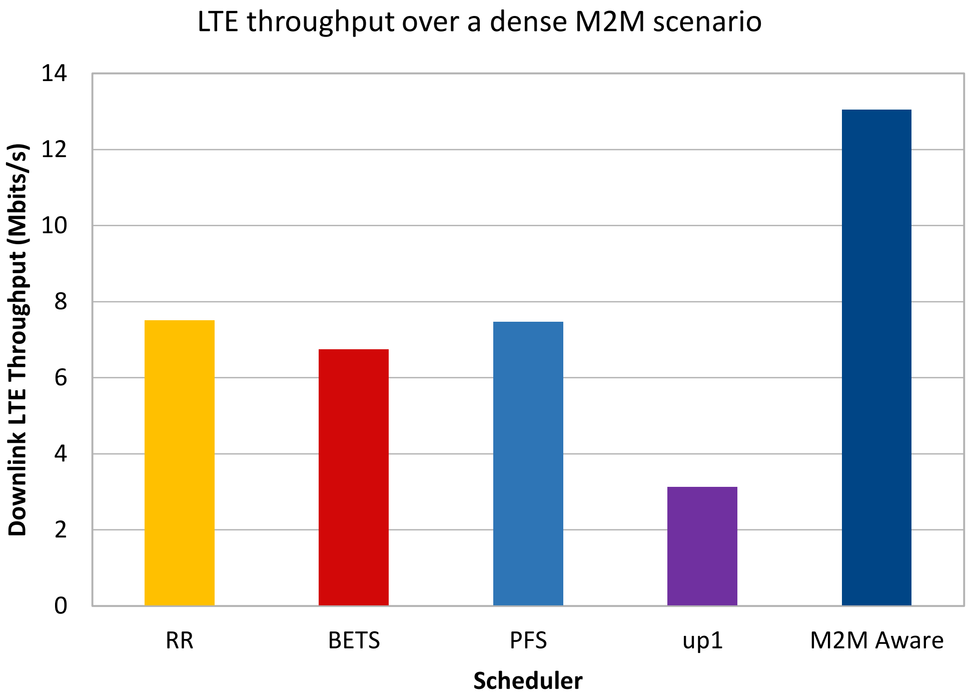

Figure 21: LTE cell throughput with different uplink schedulers

Studying the simulation results is clear to state that the current uplink scheduler are not ready and shaped to handle the M2M scenario for two main reason:

- *Full buffer addressing*: the implementation of the schedulers is made in a no queue aware way. At the TTI end the statistics that help the scheduling decision (such in PFS or BETS) are updated, that's means that for each resource block the scheduling policies will give the same results. That's totally normal if Full buffer traffic is studied but due to the small packets, of the M2M devices and the ACK packets from the LTE ones, a bad utilization of the channel is performed.

- *Uncurrelation between up and down schedulers*: The nowadays traffic is carried by TCP flows, it mean that a correlation between the scheduling policies of ad downlink scheduler and an uplink one should be exploited. Moreover the flows of the ACK packets in both the directions has to be prioritized to help devices to build their transmitting queue.

Despite the bad literature scheduler results the M2M Aware scheduler, presented in this paper, seams able, due to his channel resources utilizations, to handle the M2M dense scenario with a lost in throughput equal to 1.486 Mbits/s, compared to the 8.272 Mbits/s from the PFS Scheduler. To complete the study the M2M Aware scheduler behaviour is shown in different M2M scenario and compared to the reference one, Figure 22.

Number of M2M	0	1	5	10	25	50	75	100
Cell Throughput (Mbits/s)	15.495	15.4743	15.3739	15.2938	15.1536	14.9303	14.7336	14.009

Table 6: M2MAware Scheduler in a dense machine scenario

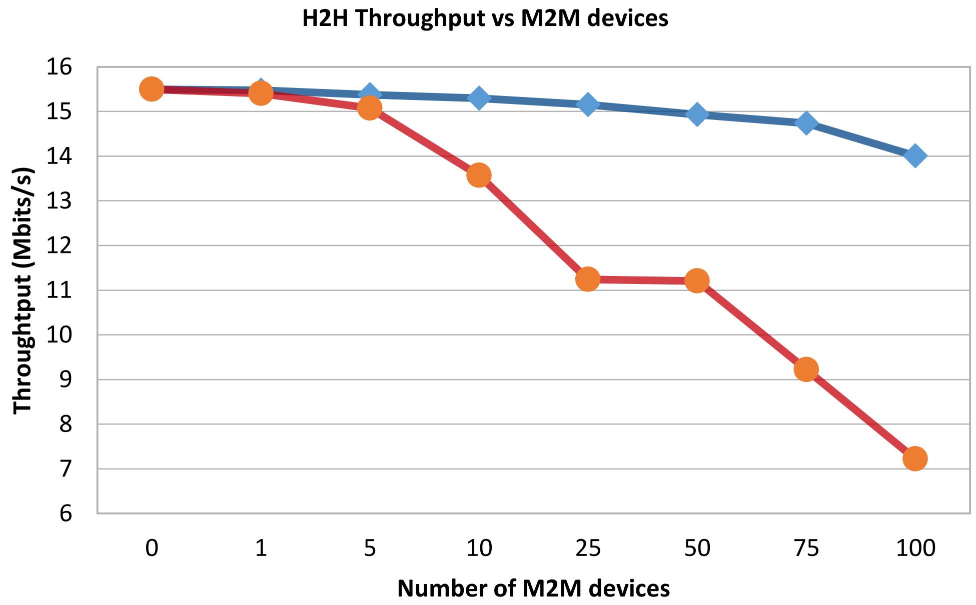

Figure 22: M2MAware Scheduler (blue line) versus PfsScheduler (red line) in machine dense scenario

Resource Blocks Utilization

Along the main 100 M2M simulations RBU data were logged. Thanks to this study is easy to state that the literature approach leads to a waste of Resource Blocks utilization from the users.

- Downlink utilization

Scheduler	LTE RB Utilization (%)	M2M RB Utilization (%)
M2MAware	89.875	30.719
PfsScheduler	97.535	4.35
BET Scheduler	91.743	3.205

Uplink #1 Scheduler	90.032	30.425

Table 7: Downlink RBU for different schedulers

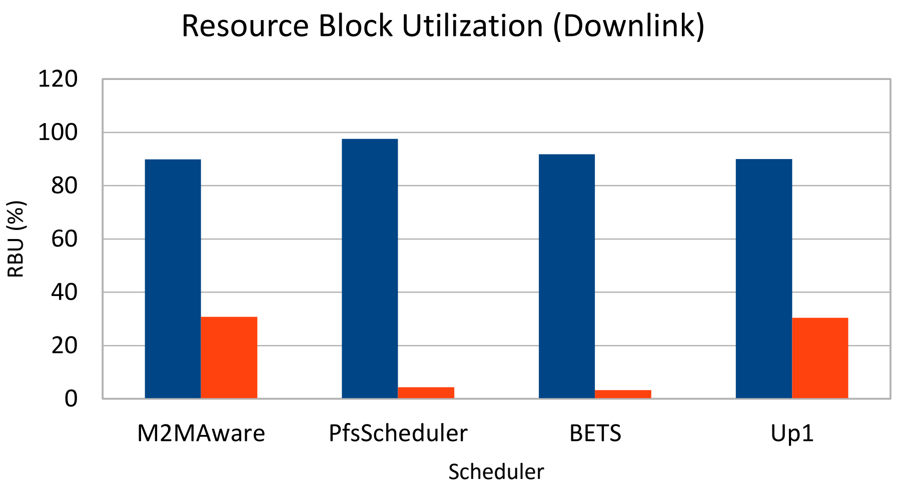

Figure 23: Downlink RBU for different schedulers (blue LTE utilization, red M2M utilization)

- Uplink utilization

Scheduler	LTE RB Utilization (%)	M2M RB Utilization (%)
M2MAware	75.589	91.352
PfsScheduler	12.481	55.214
BET Scheduler	9.181	11.561
Uplink #1 Scheduler	11.409	26.032

Table 8: Uplink RBU for different schedulers

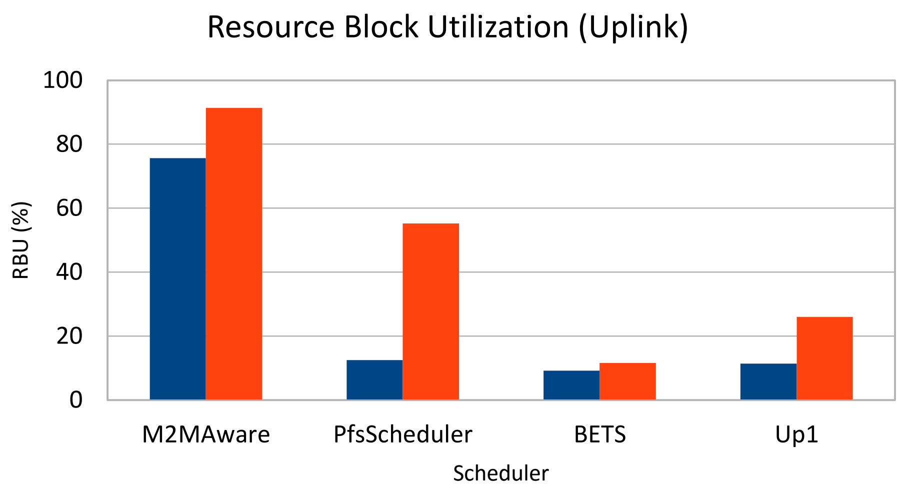

Figure 24: Uplink RBU for different schedulers (blue LTE utilization, red M2M utilization)

In Figure 23 and Figure 24 are shown the RBs utilizations both for downlink and the uplink. While is shown that with the literature schedulers the RBU is low in both uplink and downlink, is stated that thanks to the M2M Aware scheduler is possible to raise the machine-to-machine RBU from 55.2% to 91.3% for M2Ms and from 12.5% to 75.6% for LTE users.

While in the downlink is worthy to underline that the maximum M2Ms resource utilization achievable is 30.7 %, 52 bytes ACK messages, since the maximum scheduler granularity is 1 RB per user.

Round Trip Time

The Round Trip Time of the LTE TCP flows is evaluated as follow: the least scheduled LTE user is selected and thought the Wireshark analysis its ACKs Round trip times are collected and shown thought the Empirical cumulative distribution function (ECDF).

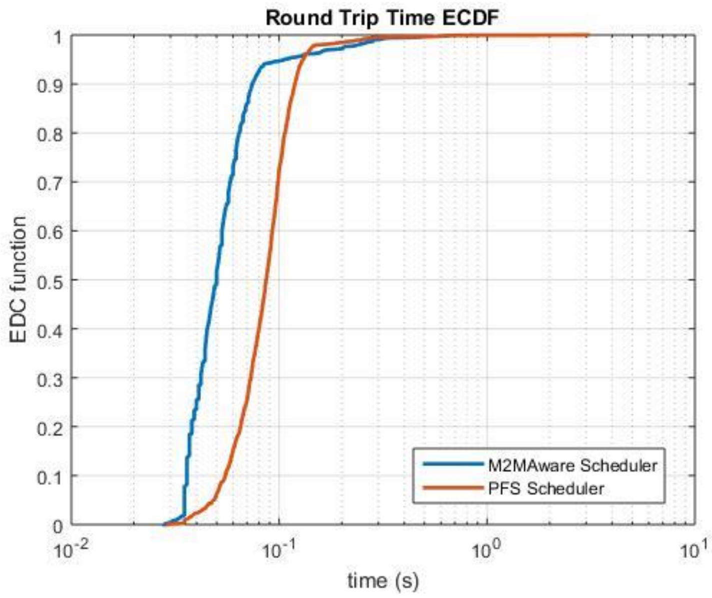

Figure 25: RTT evaluation

Thanks to the M2M scheduler allows the LTE packets to be scheduler first both in uplink and in downlink binging a better RRT behaviour respect to the standard PFS scheduler.

Conclusions

The work carried during this thesis project underline the necessity to further study and analyse the world of the M2M communications since the trends are clear about their grow. The literature schedulers are not enough to handle the massive data flow that will flood the network and more over is clear that the common approach to separate a scheduler between uplink and downlink cannot be applied anymore but both have to work along in the order to follow the data flows and avoid network congestions. Along to that a new study and standardisations has to be taken in place to find a LTE devices centric fairness index, to overcome the unreliability of the Jain's fairness scheme for such scenario. The M2M Aware Scheduler here presented seams having the right chances to handle a stressful M2M load. In future works could be exploiting it through the classification of M2M users in sets with different QoS (delay) constraints.

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