University of Western Macedonia

Department of Informatics and Telecommunications Engineering

# Development of a novel, adaptive and effective scheduling protocol for LTE wireless networks

**Isaak Georgiadis**

AEM: 300

Supervisor: Assistant Professor

Panagiotis Sarigiannidis

5 July, 2016

# Abstract

Mobile communications systems revolutionized the way people communicate, joining together communications and mobility. A long way in a remarkably short time has been achieved in the history of wireless communications. Evolution of wireless access technologies is about to reach its fifth generation (5G). Looking in the past, wireless access technologies have followed different evolutionary paths aimed at unified target: performance and efficiency in high mobile environment. The first generation (1G) has fulfilled the basic mobile voice, while the second generation (2G) has introduced capacity and coverage. This is followed by the third generation (3G), which promotes higher data delivery rates to open the gates for truly "mobile broadband" experience, which will be further realized by the fourth generation (4G).

4G mobile systems focus on seamlessly integrating the existing wireless technologies including wireless LAN and Bluetooth. Moreover they support comprehensive and personalized services, providing stable system performance and Quality of Service (QoS).

This dissertation introduces an efficient adaptive scheduling protocol for LTE wireless networks, which endeavors to provide fairness in the bandwidth allocation process. Moreover, state-of-the-art scheduling techniques are examined, implemented and compared to the proposed scheme in terms of mean packet delay, jitter and throughput. Throughout the implementation procedure, Matlab programming language was employed. Furthermore, Event-Driven programming was adopted. The evaluation results demonstrate that the proposed scheme is capable of supporting a better data delivery service compared to the conventional algorithms.

# Περίληψη

Τα κινητά συστήματα επικοινωνιών έφεραν επάνασταση στον τρόπο επικοινωνίας των ανθρώπων, παντρεύοντας την επικοινωνία με την κινητικότητα. Παράλληλα, η επίτευξη αυτού του σημαντικού βήματος πραγματοποιήθηκε σε εξαιρετικά σύντομο χρονικό διάστημα για την ιστορία των ασύρματων δικτύων, με την τεχνολογία πέμπτης γενιάς (5G) να είναι προ των πυλών. Διατρέχοντας στο παρελθόν, οι τεχνολογίες ασύρματης πρόσβασης ακολούθησαν διαφορετικές πορείες εξέλιξης αποσκοπώντας σε ενιαίο στόχο : την απόδοση και αποτελεσματικότητα σε συνδυασμό με την υψηλή κινητικότητα του χρήστη. Η πρώτης γενιάς τεχνολογία (1G) εδραίωσε την κινητή τηλεφωνία, καθώς η δεύτερη γενιά (2G) βελτιστοποίησε την χωριτικότητα και κάλυψη του δικτύου. Επακολούθησε η τρίτη γενιά (3G), η οποία επέφερε υψηλές ταχύτητες δεδομένων εισάγοντας την έννοια των κινητών ευρυζωνικών δικτύων, ενώ εν συνεχεία η τέταρτη γενιά τα καθιέρωσε.

Τα δίκτυα τέταρτης γενιάς βασίζονται στην απρόσκοπτη διαλειτουργικότητα των υφιστάμενων ασύρματων τεχνολογιών, συμπεριλαμβανομένων των ασύρματων τοπικών δικτύων και Bluetooth. Επιπλέον υποστηρίζουν μία ευρεία γκάμα υπηρεσιών, παρέχοντας σταθερή απόδοση και ποιότητα.

Στα πλαίσια αυτής της διπλωματικής παρουσιάζεται ένα καινοτόμο, προσαρμοστικό και αποδοτικό πρωτόκολλο χρονο-προγραμματισμού για ασύρματα δίκτυα LTE, που κατανέμει δίκαια το διατιθέμενο εύρος ζώνης στους χρήστες του δικτύου. Παράληλλα εξετάζεται η αποδοτικότητα αυτού του πρωτοκόλλου συγκριτικά με τους ισχύοντες αλγορίθμους δρομολόγησης ως προς την καθυστέρηση, την διακύμανση της καθυστέρησης και τη ρυθμαπόδοση. Καθ’ όλη τη διαδικασία υλοποίησης των παραπάνω αλγορίθμων δρομολόγησης χρησιμοποιήθηκε η γλώσσα προγραμματισμού Matlab, ενώ παράλληλα ιυοθετήθηκε η τεχνική του γεγονοδηγούμενου προγραμματισμού. Τέλος, τα αποτελέσματα της αξιολόγησης επιδεικνύουν ότι το προτεινόμενο πρωτόκολλο είναι αποδοτικότερο.

# Acknowledgements

Foremost, I would like to express my sincere gratitude to my supervisor Assistant Professor of the Department of Informatics and Telecommunications Engineering at the University of Western Macedonia, Dr. Panagiotis Sarigiannidis for the useful comments, remarks and engagement through the implementation process of my diploma thesis.

Last but not least, I want to thank my parents, family and friends who gave me motivation and encouraged me during the whole time period of my dissertation.

# Contents

# List of Figures

# List of Tables

# List of Abbreviations

| | |
|---|---|
| **1G** | First Generation Wireless Systems |
| **1xEV-DO** | 1x Evolution for Data Optimized |
| **2G** | Second Generation Wireless Systems |
| **3G** | Third Generation Wireless Systems |
| **3GPP** | Third Generation Partnership Project |
| **3GPP2** | Third Generation Partnership Project 2 |
| **4G** | Fourth Generation Wireless Systems |
| **BPSK** | Binary Phase Shift Keying |
| **BW** | Bandwidth |
| **C-RNTI** | Cell Radio Network Temporary Identity |
| **CDMA** | Code Division Multiple Access |
| **CFI** | Channel Format Indicator |
| **CP** | Cyclic Prefix |
| **CQI** | Channel Quality Indicator |
| **CRC** | Cyclic Redundancy Check |
| **CSI** | Channel State Information |
| **DCI** | Downlink Control Information |
| **DFT** | Discrete Fourier Transform |
| **DL** | Downlink |
| **DS** | Dynamic Scheduling |
| **E-UTRA** | Evolved UMTS Terrestrial Radio Access |
| **E-UTRAN** | Evolved UMTS Terrestrial Radio Access Network |
| **eNB** | E-UTRAN Node B |
| **EPC** | Evolved Packet Core |
| **EPS** | Evolved Packet System |
| **EV-DO** | Evolution for Data Optimized |
| **FDD** | Frequency Division Duplex |
| **FDM** | Frequency Division Multiplexing |
| **FDMA** | Frequency Division Multiple Access |
| **FFT** | Fast Fourier Transform |

| | |
|---|---|
| **GI** | Guard Interval |
| **GP** | Guard Period |
| **GPRS** | General Packet Radio Service |
| **GSM** | Global System for Mobile Communication |
| **GW** | Gateway |
| **H-ARQ** | Hybrid ARQ |
| **HDTV** | High Definition TV |
| **HSDPA** | High Speed Downlink Packet Access |
| **HSPA** | High Speed Packet Access |
| **HSS** | Home Subscriber Server |
| **HSUPA** | High Speed Uplink Packet Access |
| **IDFT** | Inverse Discrete Fourier Transform |
| **IEEE** | Institute of Electrical and Electronics Engineers |
| **IFFT** | Inverse Fast Fourier Transform |
| **IMEI** | International Mobile Equipment Identity |
| **IMS** | IP Multimedia Subsystem |
| **IMT** | International Mobile Telecommunication |
| **IP** | Internet Protocol |
| **ISI** | Inter-Symbol Interference |
| **ITU** | International Telecommunication Union |
| **kbps** | kilo-bits per second |
| **KHz** | Kilo Hertz |
| **LTE** | Long Term Evolution |
| **MAC** | Medium Access Control |
| **MBMS** | Multimedia Broadcast Multicast Service |
| **MBSFN** | MBMS Single Frequency Network |
| **MCH** | Multicast Channel |
| **ME** | Mobile Equipment |
| **MHz** | Mega Hertz |
| **MIB** | Master Information Block |
| **MIMO** | Multiple Input Multiple Output |
| **MME** | Mobility Management Entity |

| | |
|---|---|
| **MMS** | Multimedia Messaging Service |
| **MS** | Mobile Station |
| **MSC** | Mobile Switching Center |
| **OFDM** | Orthogonal Frequency Division Multiplexing |
| **OFDMA** | Orthogonal Frequency Division Multiple Access |
| **P-GW** | PDN Gateway |
| **P-SCH** | Primary Synchronization Channel |
| **PBCH** | Physical Broadcast Channel |
| **PCFICH** | Physical Control Format Indicator Channel |
| **PDCCH** | Physical Dedicated Control Channel |
| **PDN** | Packet Data Network |
| **PDSCH** | Physical Downlink Shared Channel |
| **PDU** | Protocol Data Unit |
| **PHICH** | Physical Hybrid ARQ Indicator Channel |
| **PHY** | Physical Layer |
| **PMCH** | Physical Multicast Channel |
| **PRACH** | Physical Random Access Channel |
| **PRB** | Physical Resource Block |
| **P-RNTI** | Paging Radio Network Temporary Identity |
| **PS** | Persistent Scheduling |
| **PSTN** | Public Switched Telephone Network |
| **PUCCH** | Physical Uplink Control Channel |
| **PUSCH** | Physical Uplink Shared Channel |
| **QAM** | Quadrature Amplitude Modulation |
| **QCI** | QoS Class Identifiers |
| **QoS** | Quality of Service |
| **QPSK** | Quadrature Phase Shift Keying |
| **RACH** | Random Access Channel |
| **RAN** | Radio Access Network |
| **RAT** | Radio Access Technology |
| **RB** | Resource Block |
| **RF** | Radio Frequency |

| | |
|---|---|
| **RLC** | Radio Link Control |
| **RRC** | Radio Resource Control |
| **RRM** | Radio Resource Management |
| **S-GW** | Serving Gateway |
| **S1-U** | S1 - User Plane |
| **SAE** | System Architecture Evolution |
| **SC** | Single Carrier |
| **SC-FDMA** | Single Carrier - Frequency Division Multiple Access |
| **SDU** | Service Data Unit |
| **SG** | Signaling Gateway |
| **SI** | System Information |
| **SIB** | System Information Block |
| **SIP** | Session Initiation Protocol |
| **SMS** | Short Message Service |
| **SPS** | Semi-Persistent Scheduling |
| **SRNC** | Serving Radio Network Controller |
| **TDD** | Time Division Duplex |
| **TTI** | Transmission Time Interval |
| **UDP** | User Datagram Protocol |
| **UE** | User Equipment |
| **UL** | Uplink |
| **UL-SCH** | Uplink Shared Channel |
| **UMTS** | Universal Mobile Telecommunications System |
| **UpPTS** | Uplink Pilot Time Slot |
| **VoIP** | Voice over Internet Protocol |
| **VoLTE** | Voice over Long Term Evolution |
| **WCDMA** | Wideband Code Division Multiple Access |
| **WiMAX** | Worldwide Interoperability for Microwave Access |
| **WLAN** | Wireless Local Area Networks |

# 1. Evolution of Mobile Technology from 0G to 4G.

At the early stage of 1970's, the mobile wireless industry made its first steps on technology creation and innovation. A few years later the cellular communication industry witnessed explosive growth, increasing its subscribers by approximately 40% per year. Surveys have shown that a new wireless subscriber signs up every 2.5 seconds.

By the 1990's, the average mobile subscriptions was 2%, while 10 years later it became 39%, leading to 92% of the average mobile subscription by 2010 (Figure 1). The rapid worldwide growth of communications in this area is statistically indubitable, proving its robustness and viability.

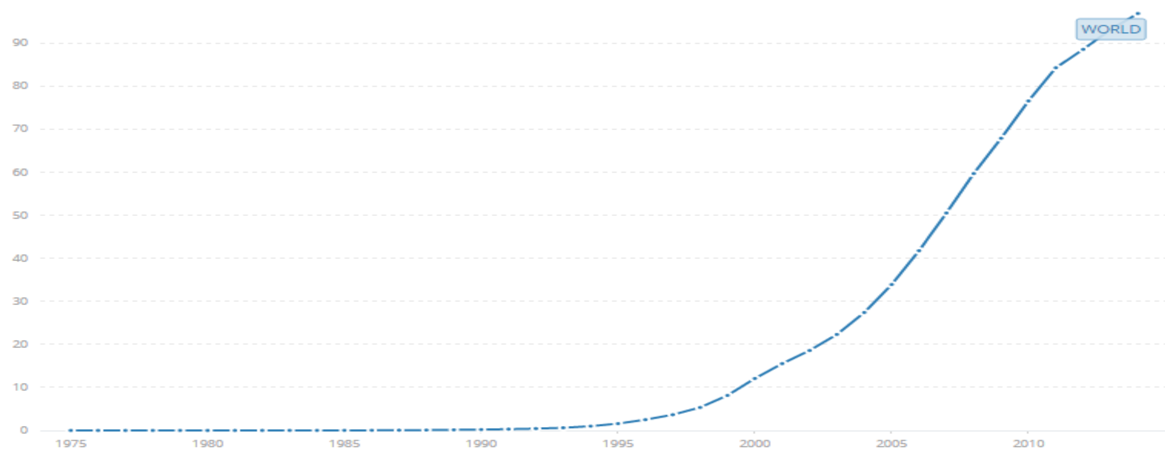
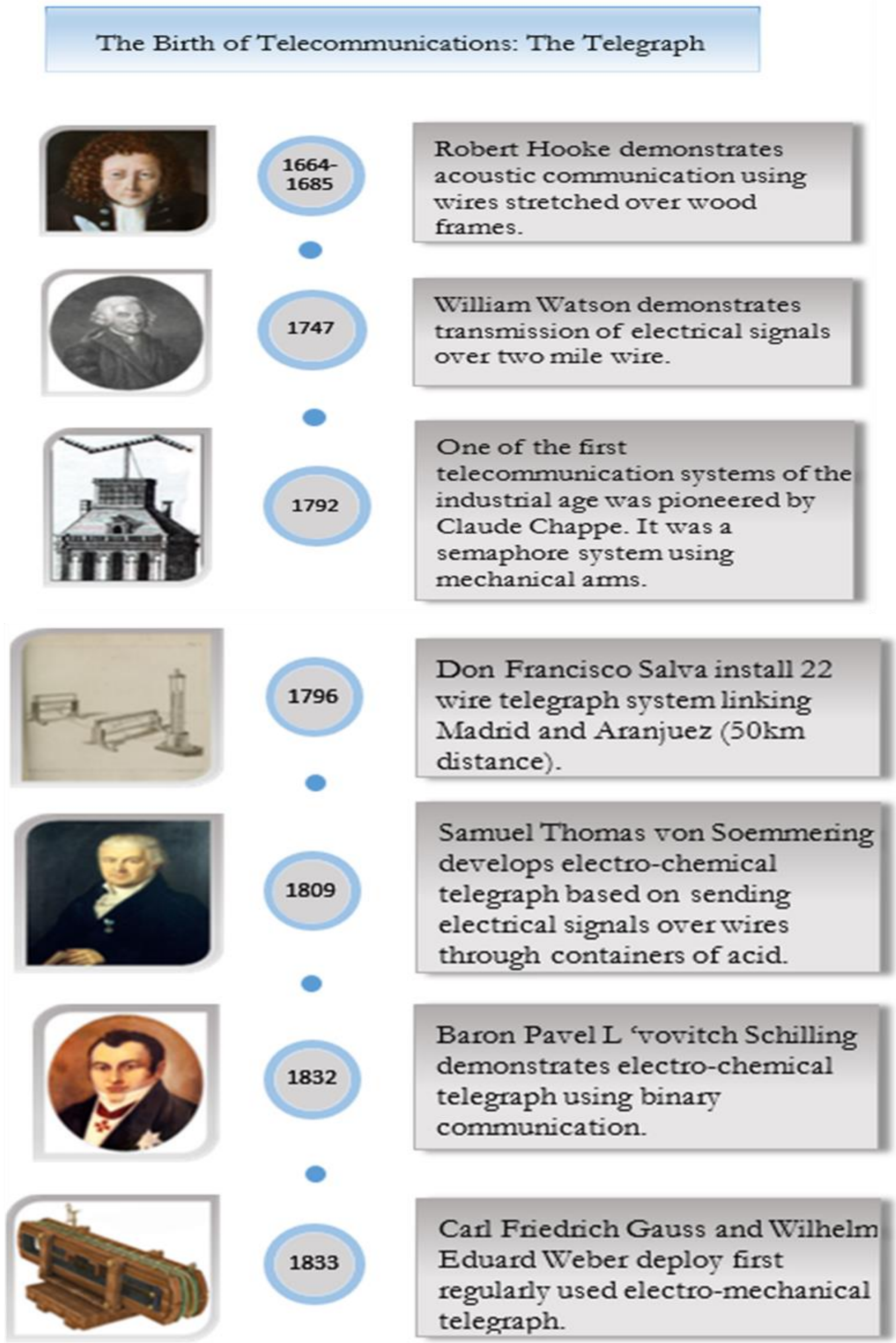
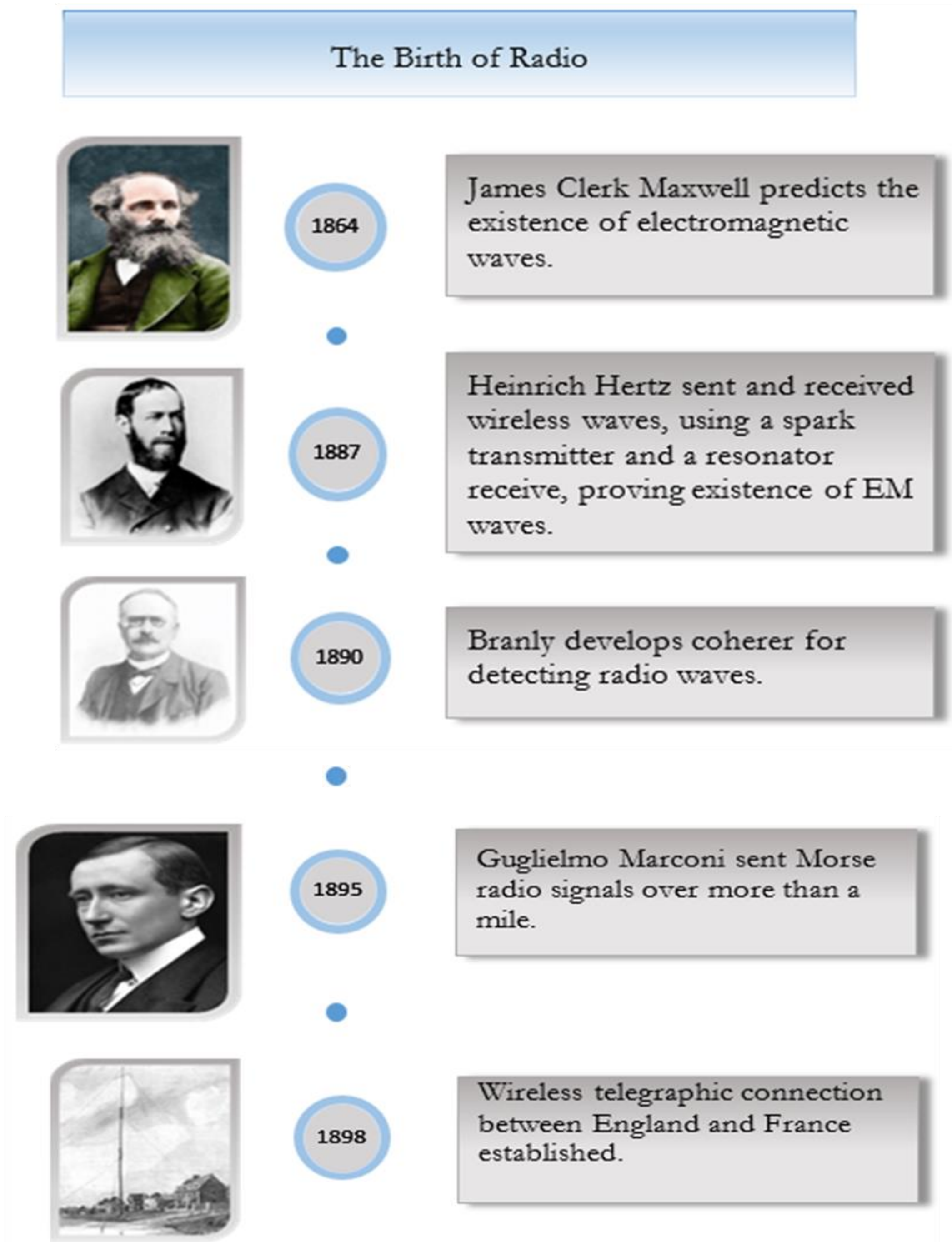


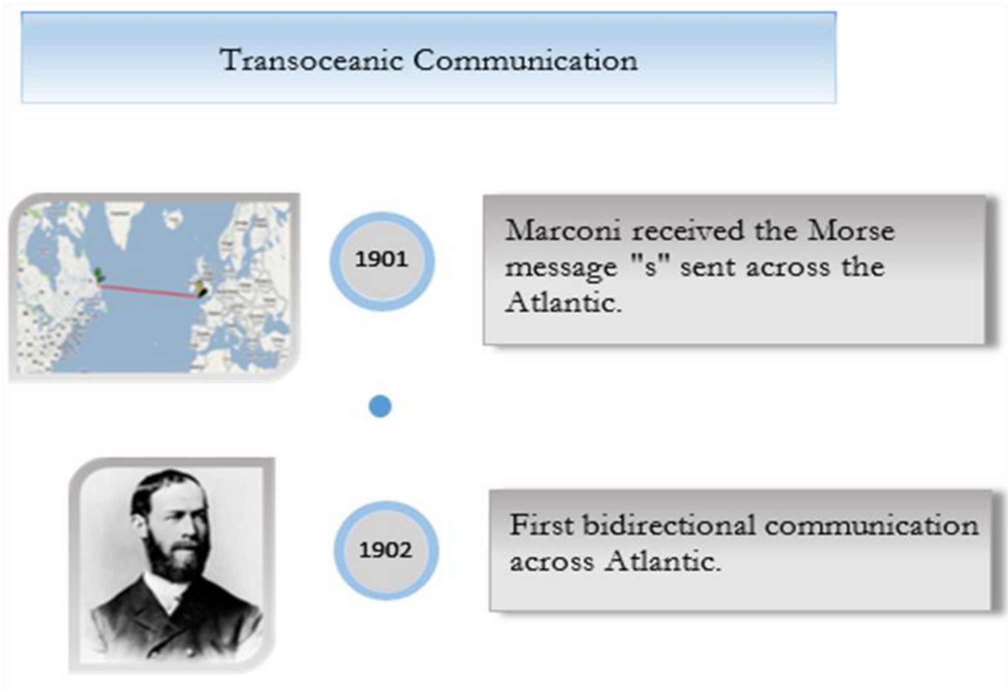

***Figure 1**: Mobile Cellular Subscriptions (per 100 people) [1]*

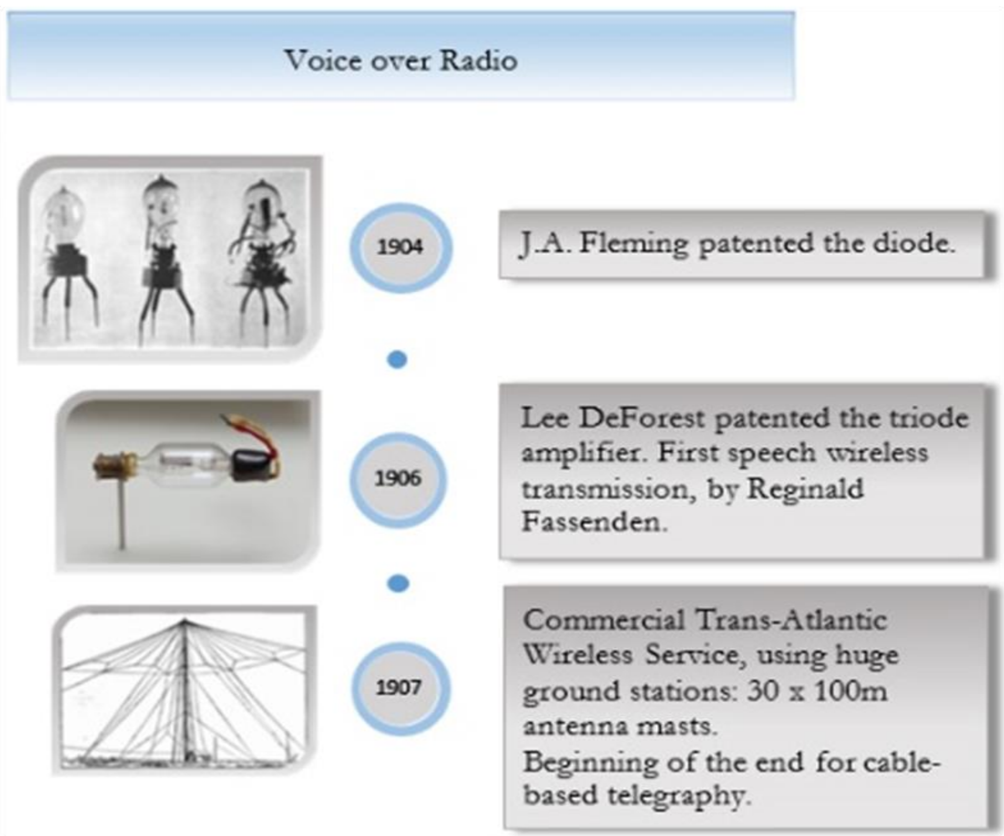
Wireless communication is nothing else but the transfer of information without the use of enhanced electrical conductors or "wires", regardless the distance. The first recorded wireless communication goes back to 1880, when Alexander Graham Bell and Charles Sumner Tainter invented and patented the photophone, a telephone that conducted audio conversations wirelessly over modulated light beams, but this was only the start. A few years later, in 1895, Guglielmo Marconi transmitted the three-dot Morse code for the letter 'S' over a distance of three kilometers using electromagnetic waves. Although, the true capabilities of the electromagnetic waves were never fully-understood until 12th December 1901, where Marconi managed to demonstrate that transatlantic wireless communication is feasible.

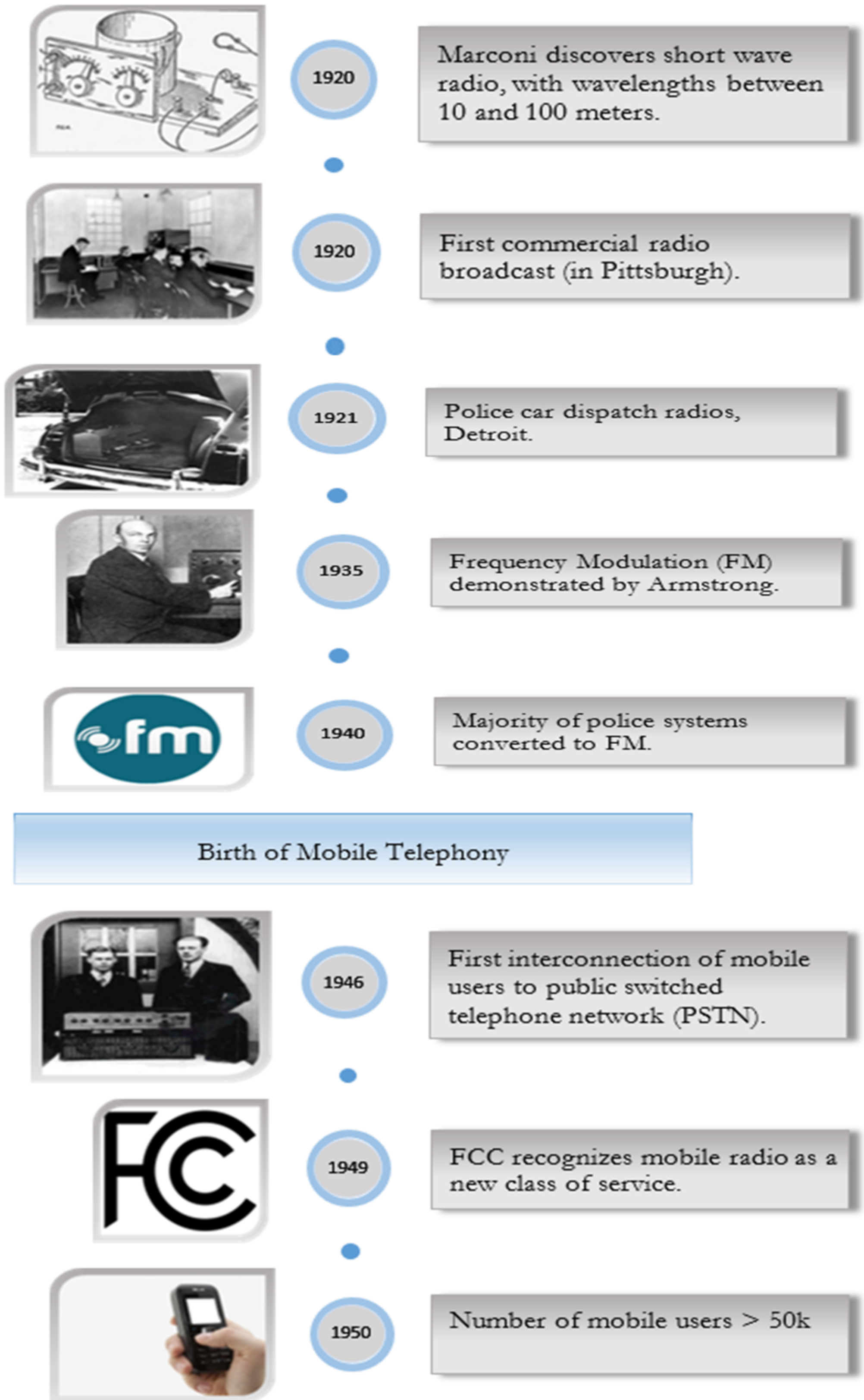
Milestones in the History of Wireless technology [2]:


The Birth of Telecommunications: The Telegraph
1664-1685
Robert Hooke demonstrates acoustic communication using wires stretched over wood frames.
1747
William Watson demonstrates transmission of electrical signals over two mile wire.
1792
One of the first telecommunication systems of the industrial age was pioneered by Claude Chappe. It was a semaphore system using mechanical arms.
1796
Don Francisco Salva install 22 wire telegraph system linking Madrid and Aranjuez (50km distance).
1809
Samuel Thomas von Soemmering develops electro-chemical telegraph based on sending electrical signals over wires through containers of acid.
1832
Baron Pavel L 'vovitch Schilling demonstrates electro-chemical telegraph using binary communication.
1833
Carl Friedrich Gauss and Wilhelm Eduard Weber deploy first regularly used electro-mechanical telegraph.


The Birth of Radio
1864
James Clerk Maxwell predicts the existence of electromagnetic waves.
1887
Heinrich Hertz sent and received wireless waves, using a spark transmitter and a resonator receive, proving existence of EM waves.
1890
Branly develops coherer for detecting radio waves.
1895
Guglielmo Marconi sent Morse radio signals over more than a mile.
1898
Wireless telegraphic connection between England and France established.


Transocenic Communication
1901
Marconi received the Morse message "s" sent across the Atlantic.
1902
First bidirectional communication across Atlantic.


Voice over Radio
1904
J.A. Fleming patented the diode.
1906
Lee DeForest patented the triode amplifier. First speech wireless transmission, by Reginald Fassenden.
1907
Commercial Trans-Atlantic Wireless Service, using huge ground stations: 30 x 100m antenna masts.
Beginning of the end for cable-based telegraphy.


1920
Marconi discovers short wave radio, with wavelengths between 10 and 100 meters.
1920
First commercial radio broadcast (in Pittsburgh).
1921
Police car dispatch radios, Detroit.
1935
Frequency Modulation (FM) demonstrated by Armstrong.
fm
1940
Majority of police systems converted to FM.
Birth of Mobile Telephony
1946
First interconnection of mobile users to public switched telephone network (PSTN).
FC
1949
FCC recognizes mobile radio as a new class of service.
1950
Number of mobile users > 50k

## 1.1 0G “The Beginning”

“The most important step of all, is the first step”

0G | 0.5G | 1G | 2G | 2.5G | 2.75G | 3G | 3.5G | 3.75G | 3.9G | 4G

### 1.1.1 Introduction

0G refers to pre-cellular mobile telephony technology that became available just after World War II. The predecessors of 1G were mainly used in cars [3].

### 1.1.2 0G Technologies

Technologies used in 0G systems included PTT (Push to Talk), MTS (Mobile Telephone System), IMTS (Improved Mobile Telephone Service), AMTS (Advanced Mobile Telephone System), OLT (Norwegian for Offentlig Landmobil Telefoni, Public Land Mobile Telephony) and MTD (Swedish abbreviation for Mobile Telephony system D).

***Table 1.*** 0G Technologies

| PTT | MTS | IMTS | AMTS | OLT | MTD |
|---|---|---|---|---|---|
| PTT also known as “Press to transmit”, a method of conversing on half duplex communication lines including two way radio without needing an existing connection. PTT can be complemented with fixed PC applications acting as PTT clients | MTS system was operator assisted in both directions in the case of one direction was called by the public switch telephone network (PSTN). If an outbound call took place, the caller would have to go through a mobile operator. | IMTS units would produce a dial tone when the receiver was lifted from the cradle and this way seemed more like a landline telephone than a cellular handset. IMTS covered an area of 40-60 miles in diameter, had 11-12-13 radio channel in larger cities while rural | AMTS operated on 900 MHz band and it overcame all the difficulties occurred from IMTS. | OLT was first land mobile telephone network in Norwegh. It operated on 160 MHz VHF band using frequency modulation on 160 -162 MHz for the mobile unit and 168-170 MHz for | MTD featured fixed wireless service with high speed internet connection without the need of telephone line. It offered “always on” internet access. |

| | | | | | |
|---|---|---|---|---|---|
| connected to mobile operator via secured internet links. | Subsequently, the operator would ask for the callees' mobile number in order to place the respective call. | stations had one or two channels. | | the base stations. | |

### 1.1.3 0.5G


0.5G is a group of technologies with improved features compared to the basic 0G technologies. The 0.5G mobile telephones were usually mounted in cars or trucks, though briefcase models. Typically, the transceiver (transmitter-receiver) was mounted in the vehicle trunk and attached to the "head" (dial, display, and handset) mounted near the driver seat. They were sold through various outlets, including two-way radio dealers. The primary users were loggers, construction foremen, realtors, and celebrities, for basic voice communication.

Early examples for this technology are shown in Table 2.

***Table 2.*** 0.5G Technologies

| ARP | B-Netz |
|---|---|
| The Autoradiopuhelin (ARP) launched in 1971 in Finland as the country's first public commercial mobile phone network. | The B-Netz launched in 1972 at Germany as the second public commercial mobile phone network of the country (but the first one that did not require human operators anymore to connect calls). |

It is noteworthy that if you could carry an 18 kg mobile cellular phone battery, you could have a portable phone with up to 30–60 min of talk time [4].

# 1.2 1G "The Analog Voice generation"

"1G established seamless mobile connectivity introducing mobile voice services"

0G | 0.5G | **1G** | 2G | 2.5G | 2.75G | 3G | 3.5G | 3.75G | 3.9G | 4G

## 1.2.1 Introduction

In 1980 the mobile cellular era had officially started, and since then mobile communications have undergone significant changes and experienced enormous growth.

This generation was the first, and simultaneously the last, which used analog transmission for speech services. That means, that the voice itself during a call was modulated to higher frequency (typically 150MHz and up), contrary to the next generations where the voice was encoded to digital signals. On the other hand, 1G used digital signaling to connect the radio towers to the rest of the telephone system in a similar manner to the next generations.

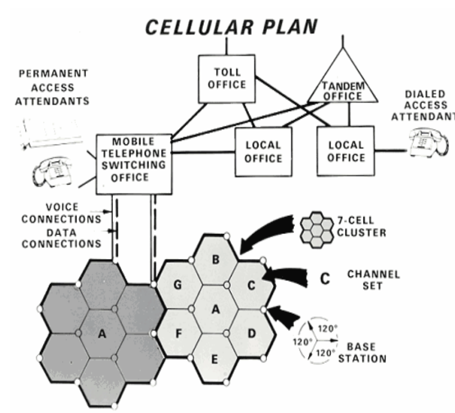



*Figure 2.* AMPS Network Architecture

The phone call establishment procedure of 1G most utilized technology, the AMPS (Advanced Mobile Phone System), is the following: When a phone is turned on, it scans for control signals from base stations. It sends this information to the BS with strongest control signal and the BS passes this information to MTS (Mobile Telephone Switching Station) as a packet. The subscriber initiates a call by keying in a phone number and pressing the send key. The MTS verifies the number and authorizes the user. MTS issues a message to the user's cell phone indicating send and receive traffic channels. MTS sends a ringing signal to the called party. Party answers; MTS establishes the circuit and initiates billing information. Either party hangs up; MTS releases the circuit, frees the channels, and completes billing.

## 1.2.2 1G Technologies

The first operational cellular system in the world was built in 1979 by Nippon Telephone and Telegraph (NTT) in Tokyo, which within five years became the first nationwide 1G network, due to its rapid expansion [5]. Two years later the cellular epoch reached Europe, with Nordic Mobile Telephones (NMT) and Total Access Communication Systems (TACS) technologies. Meanwhile, America utilized the Advanced Mobile Phone System (AMPS) technology.

***Table 3***. 1G Technologies

| NMT | AMPS | TACS |
|---|---|---|
| **NMT** is the first full automatic cellular phone system. It is an FDMA-based technology that came in service in 1981 as a response to the increasing congestion and heavy requirements of the manual mobile phone network. The two variants of NMT exists-NMT-450 and NMT-900. The numbers indicate the frequency band use respectively. NMT-900 had more channels than NMT-450. The cell size ranges from 2-30 km. NMT phones used full duplex transmission allowing for simultaneous receiving and transmission of voice. It had automatic switching and hand over of the call built from the beginning. The main problem here is that the voice was not fully encrypted. | **AMPS** was an analog mobile phone system standard develops by Bell Labs, introduced in America in 1983. It was a first generation cellular technology that used separate channels for each conversation. It was different from older systems by “back-end” call setup functionality which allowed a larger numbers of phones to be supported over a geographical area. In 1983, a total of 40 MHz of spectrum in the 800 MHz band was allocated by FCC for AMPS. In 1989, extra 10 MHz had been allocated to AMPS due to increase of cellular system capacity. For AMPS, an uplink frequency is 824-849 MHz and downlink frequency is 869-894 MHz. Each AMPS frequency is 30 KHz wide. | **TACS** was first used in UK in 1985. It was based on the AMPS technology. The TACS system was implemented in the 800–900 MHz band. After its implementation, this technology was spread to Europe, China, Singapore, Hong Kong and Japan. |

## 1.2.3 Advantages & Disadvantages of 1G

1G established the foundation of mobile, introducing the following notions:

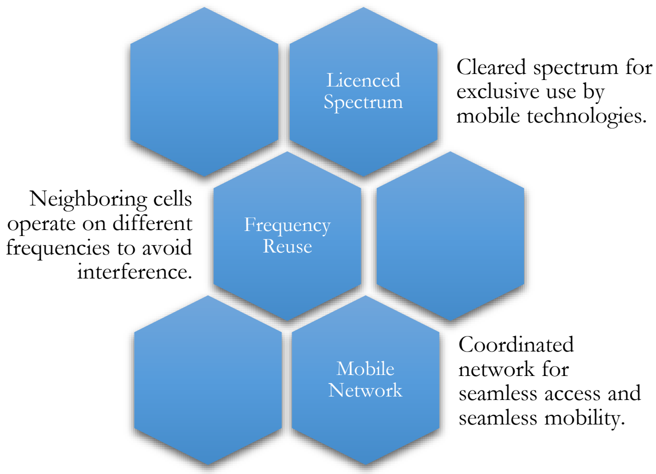



***Figure 3.*** Advantages of 1G

The key idea of 1G cellular networks is that the geographical area is divided into cells (typically 10-25 km), each served by a "base station", as shown in Figure 3. Cells are small so that frequency reuse can be exploited in nearby (but not adjacent) cells. This allows many more users to be supported in a given area. For instance, compared to IMTS, AMPS can support 5 to 10 times more users in the same 100-mile area by dividing the area into 20 smaller cells that reuse the same frequency ranges [6].

One of the key improvements in 1G was the invention of the microprocessor, whilst the drawbacks were many. Some of them are the following:

- No interoperability between countries.
- Limited capacity.
- Unreliable handoff.
- Poor voice links.
- Analogue signals.
- Security Flaws.
- Limited Scalability.

First of all, the most concrete drawback of 1G was the limited scalability of its devices. The size of a standard device was pretty large, while their weight was heavy. Beyond those, power inefficiency and high cost were two extra features that made the devices less user friendly.

***Figure 4.*** 1G Telephone

Capacity was an issue back in 1980's because analogue transmissions are inefficient at using limited spectrum. That inefficiency is caused firstly by the large frequency gap between users to avoid interference, and secondly because only one user is supported per channel.

Security was also a hot topic in 1G. First of all voice calls were played back in radio towers making these calls susceptible to unwanted. Anyone with an all-band radio receiver can listen in to the conversation. There are also thefts of airtime. Basically, a thief uses an all-band radio receiver connected to a computer. This computer can record 32-bit serial numbers and phone numbers of subscribers when they are calling (recall that this information is sent as a packet). The thieves can collect a large database by driving around and then go into business by reprogramming stolen phones and reselling them.

***Figure 5.*** Eavesdropping

Last but not least, the main disadvantage of first generation mobile network was the lack of interoperability between countries [7]. The reason behind this, is that different standards were used in various countries and there was no possible way to perform a vertical handoff. United States used AMPS, while NMT was used in Nordic countries (Denmark, Finland, Norway, Iceland and Sweden) as well as Switzerland, Netherlands, Eastern Europe and Russia. In Italy, the telecommunication standard used was called RMTI, UK used TACS and France used Radiocom 2000. In West Germany, South Africa and Portugal a telecommunication system known as C-450 was used.

***Table 4***. 1G Main Features

| | |
|---|---|
| Generation | 1G |
| Starts-Ends | 1970-1984 |
| Frequency | 800-900 MHz |
| Data Capacity | 2 Kbps |
| Technology | Analog wireless |
| Multiplexing | FDMA |
| Switching | Circuit |
| Service | Voice only |
| Main Network | PSTN |
| Standard | AMPS |
| Handoff | Horizontal |

# 1.3 2G "Digitalization"

"2G digital wireless technologies increased voice capacity delivering mobile to the masses"


## 1.3.1 Introduction

By the early 1990's, it was about time to move on the next generation. Advances in integrated circuit (IC) technology had made digital communications not only practical, but, actually more economical than analog technology [8].

Our entry to the digital communications era allowed us to use the spectrum much more efficiently, and thereby reduced the amount of bandwidth required for services like voice and video. With the introduction of digital encryption, security also received a mild update in such a way that only an intended receiver can receive data. Last but not least, 2G offers data rates in the range of 9.6 Kbps-14.4 Kbps and introduced data services for mobile, starting with SMS text messages.

Different approaches to 2G have been developed in the US and Europe. In the US, divergence happened because only one AMPS existed. Due to this fact, several technologies emerged to compete in 2G. On the other hand, in Europe, exactly the reverse happened – there was a convergence because there were many incompatible 1G systems with no outsiders. This caused a major interoperability problem for the users mainly because you could not use your telephone while traveling abroad. European PTT (Post, Telephone and Telegraphic) sponsored development of the now very popular GSM that uses new frequency ranges and complete digital communication.

The new systems introduced in 2G can be classified by their multiple access techniques as either Time Division Multiple Access (TDMA) or Code Division Multiple Access (CDMA). FDMA was widely used in the 1G systems is considered outdated (Figure 6).

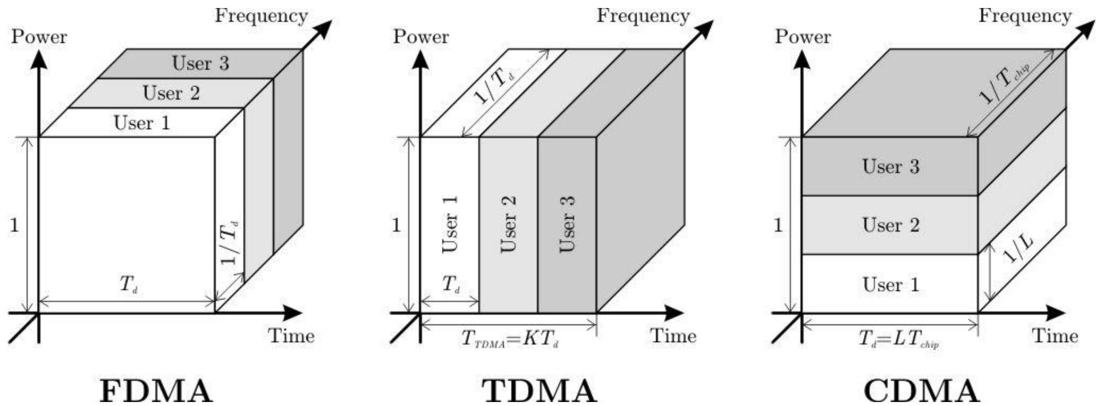



***Figure 6.*** Multiple Access Techniques of 1G and 2G *[34]*

In TDMA, several users transmit at the same frequency but in different time slots, while CDMA uses the principle of direct sequence spread-spectrum where the signals are modulated with high bandwidth spreading waveforms, called signature waveforms or codes. Although the users transmit at the same frequency and time, separation of signals is achieved because the signature waveforms have very low cross correlation.

Following, we analyze some of the differences between those two modulation techniques.

### 1.3.2 TDMA vs CDMA

Ever since the invention and commercialization of cellular systems, industry leaders and engineers had already foreseen the inevitable overcrowding of simultaneous callers because the number of accessible channels is very limited. Engineers are now using methods to remedy these problems. Examples of such methods are TDMA and CDMA. Even though they differ, they accomplish same objectives. Their primary goal is to significantly increase the quantity of concurrent users within a definite section of radio spectrum.

The debate is raging because hardware vendors have chosen sides and consequently the standardizing bodies have been lobbied hard. The primary motivation about this level of debate is that vendors want their selection to become the industry standard. Since both TDMA and

CDMA have become TIA (Telecom Industry Association) standards – IS-54 and IS-95, respectively – the debate goes on to determine which standard is better.

Technically speaking, CDMA has the following advantages and disadvantages over TDMA, shown in Table 5.

***Table 5.*** Advantages and Disadvantages of TDMA vs CDMA

**Advantages**

**Network capacity**: In CDMA, the same frequency can be reused in adjacent cells because the user signals differentiate from each other by a code. Thus frequency reuse can be very high and many more users (up to 10 times more) can be supported as compared to TDMA.

**Privacy**: Privacy is inherent in CDMA since spread spectrum modulates data to signals randomly (you cannot understand the signal unless you know the randomizing code).

**Reliability and graceful degradation**: CDMA-based networks only gradually degrade as more users access the system. This is in contrast to the sudden degradation of TDMA based systems. For example, if the channel is divided between ten users, then the eleventh user can get a busy signal in a TDMA system. This is not the case with CDMA because there is no hard division of channel capacity – CDMA can handle users as long as it can differentiate between them. In case of CDMA, the noise and interference increases gradually as more users are added because it becomes harder to differentiate between various codes.

**Frequency diversity**: CDMA uses spread spectrum, thus transmissions are spread over a larger frequency bandwidth. Consequently, frequency-dependent transmission impairments that occur in certain frequency ranges have less effect on the signal.

**Environmental**: Since existing cells can be upgraded to handle more users, the need for new cell towers decreases.

**Disadvantages**

**Relatively immature**: As compared to TDMA, CDMA is a relatively new technology but it is catching up fast.

**Self-jamming**: CDMA works better if all mobile users are perfectly aligned on chip (code) boundaries. If this is not the case, then some interference can happen. This situation is better with TDMA and FDMA because time and frequency guard bands can be used to avoid the overlap.

**Soft handoff**: An advantage of CDMA is that it uses soft handoff (i.e., two cells can own a mobile user for a while before the handoff is complete). However, this requires that the mobile user acquires the new cell before it relinquishes the old – a more complex process than hard handoff used in FDMA and TDMA schemes.

The main advantage of CDMA is that the frequency reuse can be very high and many more users can be supported in a cell as compared to TDMA. Although this leads to a soft handoff that is more complicated than the hard handoff used in TDMA, the advantage of supporting more users far outweighs the disadvantage of added complexity.

### 1.3.3 2G Technologies

2G technology was commercially launched on the GSM standard in Finland by Radiolinjia, where the former Finnish Prime Minister Harri Holkeri made the first call on July 1 in 1991. The following year, the first SMS was sent and Vodafone UK and Telecom Finland signed the first international roaming agreement. After 2G was launched, the previous mobile telephone systems were retrospectively dubbed 1G.

As mentioned before, 2G technologies can be grouped by the multiple access technique used. Technologies are described in Table 6.

***Table 6***. 2G Technologies

| TDMA | | | | CDMA |
|---|---|---|---|---|
| GSM | PDC | I-den | D-AMPS (IS 136) | cdmaOne (IS-95) |
| GSM is used in more than 212 countries in the world, making international roaming very common and enabling subscribers to use their phones in many parts of the world. Today, accounts for over 80% of all subscribers around the world. These networks operate at 9.6 Kbps and are based on international standards defined by the European Telecommunications Standards Institute (ETSI). GSM has enabled the users to make use of the short message services (SMS) to any mobile network at any time. It supports 8 users per channel. | PDC or personal digital cellular technology was developed and used in JAPAN. PDC uses 25 KHz frequency, pi/4 DQPSK modulati on with 3-timeslot 11.2 kbit/s (full-rate) or 6-timeslot 5.6 kbit/s (half-rate) voice codecs. Docomo launched its first digital service of PDC in 1993. | Integrated digital enhanced network (iDEN) was developed by MOTOROLA. It enabled the mobile users to make use of complex trunked radio and mobile phones. iDEN has a frequency of about 25Khz, but only occupies 20 kHz in order to provide interference protection via guard bands. iDEN allows three or six user per mobile channel. | IS-136 is a second generation cellular phone system that functions as an evolution of the IS-54 standard. D-AMPS uses existing AMPS c hannels and allows for smooth transition between digital and analog systems in the same area. It can support 3 users per channel. | IS-95 was the first CDMA-based technology, developed by QUALCOMM. The publication date of IS-95 Revision A was in May 1995, while Revision B's was on March 1999. Like TDMA IS-136, CDMA operates in the 1900-MHz band as well as the 800 band. The analog voice signals are digitally encoded, using QPSK (Quadrature Phase Shift Keying), at 9600 bps. It supports data rates up to 14.4 Kbit/s. |

## 1.3.4 Advantages & Disadvantages of 2G

Some of the most important advantages of 2G systems over their predecessors are described in Table 7.

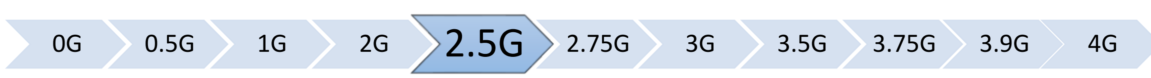
***Table 7.*** Advantages and Disadvantages of 2G

| Advantages | Disadvantages |
|---|---|
| Enabled simple data services, such as SMS and Email. | Analog has a smooth decay curve, digital a jagged steppy one. This can be both an advantage and a disadvantage. Under good conditions, digital will sound better. Under slightly worse conditions, analog will experience static, while digital has occasional dropouts. As conditions get worse, though, digital will start to completely fail, by dropping calls or being unintelligible, while analog slowly gets worse, generally holding a call longer and allowing at least a few words to get through. |
| The digital systems were designed to emit less radio power from the handsets. This meant that cells had to be smaller, so more cells had to be placed in the same amount of space. This was made possible by cell towers and related equipment getting less expensive. | While digital calls tend to be free of static and background noise, the lossy compression used by the codecs takes a toll; the range of sound that they convey is reduced. You will hear less of the tonality of someone's voice talking on a digital cell phone, but you will hear it more clearly. |
| The lower power emissions helped address health concerns. | |
| Enhanced privacy: A key digital advantage not often mentioned is that digital cellular calls are much harder to eavesdrop on by use of radio scanners, whereas 1G had no protection against eavesdropping. | |
| In terms of capacity, digital voice data can be compressed and multiplexed much more effectively than analog voice encodings through the use of various codecs, allowing more calls to be packed into the same amount of radio bandwidth. Moreover, 2G allows multiple users per radio channel, even though the spectrum gap between users remains large for interference purposes. | |
| Scalability was also a new feature for wireless generations, as digital components now cost and weight far less. Furthermore, the devices become pocket-sized. | |

*Table 8.* 2G Main Features

| | |
|---|---|
| Generation | 2G |
| Starts-Ends | 1990-2000 |
| Frequency | 850-1900 MHz(GSM)<br>825-849 MHz(CDMA) |
| Data Capacity | 10 Kbps |
| Technology | Digital wireless |
| Multiplexing | TDMA / CDMA |
| Switching | Circuit / Packet |
| Service | Voice / Data |
| Main Network | PSTN |
| Handoff | Horizontal |
| Standard | CDMA TDMA GSM |

## 1.3.5 2.5G




2.5G which stands for "second and a half generation", is a cellular wireless technology developed between its predecessor, 2G, and its successor, 3G. The move into 2.5G world began with General Packet Radio Service (GPRS). GPRS is a radio technology that adds packet-switching protocols, and introduces a new billing method where data transfer is charged per megabyte of traffic transferred, rather than connection time [9].

GPRS uses GMSK modulation (1 bit per symbol) and it is able to provide data rates from 56 kbit/s up to 115 kbit/s. It can be used for services such as:

- Wireless Application Protocol (WAP) access
- Multimedia Messaging Service (MMS)
- Email

- Web surfing

Subsequently, GPRS can be added to GSM infrastructures quite readily. It takes advantage of existing 200 kHz radio channels and does not require new radio spectrum. GPRS basically overlays a packet switching network on the existing circuit switched GSM network. This gives the user an option to use a packet-based data service. An architectural view of GPRS is presented in Figure 7. The main component of a GPRS network is the SGSN (GPRS Support Node) that receives the packet data and transfers it to the Internet or other GPRS networks. To provide GPRS services on top of GSM, the network operators need to add a few GSNs and make a software upgrade to BSCs and few other network elements. This quick upgrade capability has fueled the popularity of GPRS.

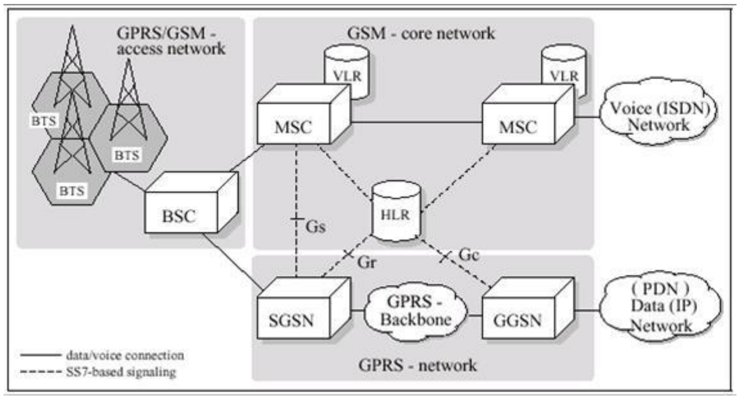



***Figure 7.*** GPRS Network Architecture

***Table 9.*** 2.5G Main Features

| | |
|---|---|
| Generation | 2.5G |
| Starts-Ends | 2000-2003 |
| Frequency | 850-1900 MHz |
| Data Capacity | 160 Kbps |
| Technology | GPRS |
| Multiplexing | TDMA / CDMA |
| Switching | Packet |
| Service | MMS / Internet |
| Main Network | GSM TDMA |
| Standard | Supported TDMA / GSM |

### 1.3.6 2.75G


2.75G introduced new methods at the physical layer, including a new form of modulation (8 PSK, 3 bits per symbol) and different ways of encoding data to protect against errors, while the higher layer protocols stayed the same [10]. 2.75G, often called EDGE (or EGPRS), is an abbreviation for Enhanced Data rates for GSM Evolution. It is a digital mobile phone technology which acts as an enhancement to its predecessor, GPRS. EDGE was invented and introduced by Cingular, which is now known as AT&T. This technology is preferred over GSM due to its flexibility to carry packet switched data and circuit switched data. Typical EDGE implementations generally do not obtain 3G data rates though, leading people to call it 2.75G.

EDGE is a pure TDMA technology that was designed to be backward compatible with the existing IS-136(D-AMPS) TDMA digital cellular phone system. This innovation was often called UWC-136. Its implementation was planned in phases by first upgrading existing IS-136 TDMA systems to IS-136+ to provide data rates of 43.2 Kbps over standard 30-kHz channels. Then IS-136+ will be upgraded to IS-136HS, which is the high-speed component of UWC-136. Eventually UWC-136 will provide packet-data services at speeds of up to 384 Kbps for wide-area coverage and up to 2 Mbps for in-building coverage. Voice communication on this system will be high-fidelity wireline quality, comparable to Integrated Services Digital Network (ISDN).

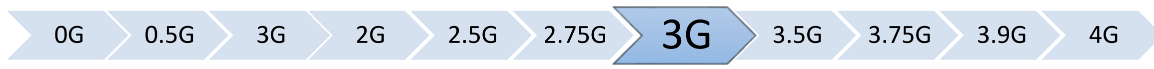
EDGE also allows the clear and fast transmission of data and information. For example, a typical text file of 40KB is transferred in only 2 seconds as compared to the transfer from GPRS technology, which is 6 seconds. Moreover, the biggest advantage of using EDGE technology is one does not need to install any additional hardware and software in order to make use of it. If a person is an ex GPRS user he can migrate to EDGE without paying any additional charges.

***Table 10***. 2.75G Main Features

| | |
|---|---|
| Generation | 2.75G |
| Starts-Ends | 2003 |
| Frequency | 850-1900 MHz |
| Data Capacity | 473 Kbps |
| Technology | EDGE |
| Multiplexing | TDMA / CDMA |
| Switching | Packet |
| Service | |
| Main Network | PSTN |
| Standard | GSM / CDMA |

## 1.4 3G "Mobile Broadband"

> "3G optimized mobile for data enabling mobile broadband services, and is evolving for faster and better connectivity"




### 1.4.1 Introduction

3G was created in an attempt to implement a ubiquitous wireless communication standard for all countries throughout the world. 3G is not one standard; it is a family of standards which can all work together [11]. All these standards shall fulfill the specifications mentioned by IMT-2000, which is the globally coordinated definition of 3G. These requirements are the following:

- Improved system capacity
- Backward compatibility with 2G systems
- Multimedia support
- High speed packet data services
- Data rates up to 2 Mbps for indoor environments, up to 384 kbps for pedestrian and urban environments and up to 144 kbps for wide area mobile environments. In other words, 3G systems mandate data rates of 144 Kbps at driving speeds, 384 Kbps for outside stationary use or walking speeds, and 2 Mbps indoors.

From the user's side, this means that 3G is capable of support a wide range of services, such as:

- Enhanced audio and video streaming
- Web and WAP browsing at higher speeds
- Video conferencing
- GPS

- IPTV
- Telemedicine

### 1.4.2 3G Technologies

The first pre-commercial 3G network was launched by NTT DoCoMo in Japan on 1998, branded as FOMA. It was first available in May 2001 as a pre-release of WCDMA technology. The first commercial launch of 3G was also by NTT DoCoMo in Japan on 1 October 2001, although it was initially somewhat limited in scope. Furthermore, the broader availability of the system was delayed by apparent concerns over its reliability.

Generally, 3G is a CDMA based generation as depicted in Table 11.

***Table 11.*** 3G Technologies

| CDMA | | |
|---|---|---|
| CDMA-2000 1xRTT(Voice) | CDMA-2000 EV-DO(Data) | UMTS (a.k.a FOMA, WCDMA) |
| It was developed by 3G PP2 as a backwards-compatible successor to second-generation cdmaOne (IS-95) set of standards and used especially in North America and South Korea. Its publication date was on October 1999. The frequency band required is a duplex pair of 1.25 MHz radio channels. 1xRTT almost doubles the capacity of IS-95 by adding 64 more traffic channels to the forward link, orthogonal to (in quadrature with) the original set of 64. It supports packet data speeds of up to 153 | CDMA2000 1xEV-DO introduced high-speed, packet-switched techniques designed for high-speed data transmissions, enabling peak data rates beyond 2 Mbps. 1xEV-DO expanded the types of services and applications available to end users, enabling carriers to broadcast more media-rich content, while users could enjoy near-wireline speeds on mobile devices. The technology has gone through several revisions from Release 0 to Revision A, Revision B, and DO-Advanced. | UMTS, short for Universal Mobile Telecommunications System, is a networking standard used throughout much of the world as an upgrade to existing GSM mobile networks. UMTS makes use of WCDMA, a technology that shares much with CDMA networks used throughout the world, though it is not compatible with them. Even though this standards pre-commercial launch goes back to 1998(on 3G's first emergence), it was not considered mature |

| | | |
|---|---|---|
| kbit/s with real world data transmission averaging 80-100 kbit/s. CDMA2000 received a handful of upgrades starting from Revision A on July 2000. After two years, on February 2002, Revision B was ready for commercial use, but that was not quite enough because on May of 2002 Revision C was ready to go. CDMA2000 Revision D was not ready until March 2004. Last but not least, CDMA2000 1X is evolving to 1X Advanced, which quadruples the capacity of today's 1X networks. | Revision A introduced enhanced capabilities such as reduced latency, QoS support and increased the peak data rate to 3.1 Mbit/s in the forward link and 1.8 Mbit/s in the reverse link. Revision B additionally offers multicarrier capabilities and provides data rates up to 4.9 Mbit/s in the DL and 1.8 Mbit/s in the UL. All revisions are completely downward compatible and are based on a channel bandwidth of 1.25 MHz operating on the same frequency bands, making EV-DO technology and test solutions requirements similar to CDMA2000 1xRTT. | enough to be used in a commercial environment until 2002. Base level UMTS networks are generally capable of the following downlink speeds:<br>Rural: 144Kbps<br>Urban: 384Kbps<br>Indoor: 1.92Mbps<br><br>UMTS was introduced in 3GPP Release 99 (R99) but it received an upgrade on Release 4 on March 2004. |

### 1.4.3 Advantages & Disadvantages of 3G

3G technologies enable network operators in order to offer users a wider range of advanced services, while achieving greater network capacity through improved spectral efficiency. In this way 3G optimized the channel capabilities by supporting larger package sizes. Some advantages of 3G systems offer over their predecessors are:

- More bandwidth, security (3G networks permit validation measures when communicating with other devices) and reliability.

- Interoperability among service providers
- Always online devices – 3G uses IP connectivity which is packet based
- Support to devices with backward compatibility with existing networks.

Some disadvantages of 3G are the following:

- The cellular infrastructure upgrade cost is really high (Base Station upgrade), as is its maintenance.
- Handsets that supported 2G could not be upgraded to 3G once it became available.
- Power consumption is high.
- From the customers point of view the expenditure for 3G network will be excessively high if they make use of the various applications of 3G.

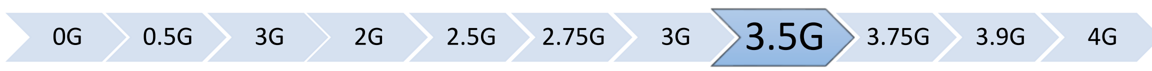
Generally, the indoor rate of 2 Mbps from 3G competes with high-speed 802.11 wireless LANs that offer data rates of 11 to 54 Mbps. The main attraction of 3G is the 384 Kbps data rate for outdoor use, as an IP-based packet-switching service over wide areas. This service can support wireless Internet access over wide geographical areas.

It is also noteworthy that 2.5G systems and beyond are based on packet switching instead of the older circuit-switching systems used in 2G. This means that in 2G cellular networks, most data communication, apart from the Short Message Service (SMS), require a circuit-switched connection in which a user must connect to a server to check email, for example. The main limitation of this approach is that the users have to be online even when they are not sending data, so they pay higher costs and network capacity is wasted.

In packet switched communications data are split into packets in which an address uniquely identifying the destination is appended. This mode of transmission, where the communication is broken into packets, allows the same data path to be shared among many users in the network. By breaking data into smaller packets that travel in parallel on different channels, the data rate can be increased significantly. For example, splitting a message into 6 packets can theoretically increase data rate six times (e.g. from 9.6 Kbps to 56 Kbps, roughly). In addition, users can stay online throughout and yet not be charged for the time spent online.

Rather, they only pay for the amount of data that they retrieve. This occurs in contrast to a circuit-switched network like the regular voice telephone network where the communication path is dedicated to the callers, thus blocking that path to other users for that period of time. Hence, although a 3G handset is, in effect, permanently connected to the network, it only uses bandwidth when needed.

### 1.4.4 3.5G




High Speed Downlink packet Access (HSDPA) is a technology that provides a smooth evolutionary path for UMTS-based 3G networks allowing for higher data transfer speeds. The introduction was done in 3GPP Release 5. HSDPA is a packet-based data service that supports a downlink speed of 14.4 Mbit/s and an uplink speed of 0.384 Mbit/s, although in real world transmissions it can achieve 8-10 Mbit/sec downlink speed. Moreover, it decreases latency as compared to 3G as well as Round Trip Time for applications and it is cheap because it functions as an upgrade to the already existing 3G systems.

This upgrades efficiency lies on the following reasons:

- Modulation: One of the keys to the operation of HSDPA is the use of an additional form of modulation. Originally W-CDMA had used only QPSK as the modulation scheme, however under the new system 16-QAM which can carry a higher data rate, but is less resilient to noise is also used when the link is sufficiently robust. The robustness of the channel and its suitability to use 16-QAM instead of QPSK is determined by analyzing information fed back about a variety of parameters. These include details of the channel physical layer conditions, power control, Quality of Service (QoS), and information specific to HSDPA.
- Fast HARQ: Fast HARQ (hybrid automatic repeat request), has also been implemented along with multi-code operation and this eliminates the need for a variable spreading factor. By using these approaches all users, whether near or far from the base station are able to receive the optimum available data rate.

- Improved scheduling: Further advances have been made in the area of scheduling. By moving more intelligence into the base station, data traffic scheduling can be achieved in a more dynamic fashion. This enables variations arising from fast fading can be accommodated and the cell is even able to allocate much of the cell capacity for a short period of time to a particular user. In this way the user is able to receive the data as fast as conditions allow.
- Additional channels: In order to be able to transport the data in the required fashion, and to provide the additional responsiveness of the system, additional channels have been added.

### 1.4.5 3.75G

After the sixth release of 3GPP, where HSUPA was introduced, the downlink speed stayed the same, while the uplink was boosted. HSUPA is the companion technology to HSDPA, so the combination of HSDPA and Enhanced UL is referred to as HSPA.

Some of the features that HSUPA introduced to the uplink direction are the following:

- Increased data rate: The use of HSUPA is able to provide a significant increase in the data rate available. It allows peak raw data rates of 5.74 Mbps.
- Lower latency: The use of HSUPA introduces a TTI of 2 ms, although a 10ms TTI was originally used and is still supported.
- Improved system capacity: In order to enable the large number of high data rate users, it has been necessary to ensure that the overall capacity when using HSUPA is higher.
- BPSK modulation: Originally only BPSK modulation, which adopted for UMTS, was used. Accordingly it did not support adaptive modulation schemes.
- Hybrid ARQ: In order to facilitate the improved performance the Hybrid ARQ (Automatic Repeat reQuest) used for HSDPA is also employed for the uplink, HSUPA.
- Fast Packet Scheduling: In order to reduce latency, fast packet scheduling has been adopted again for the uplink as for the downlink, although the implementation is slightly different.

Concluding, HSUPA provides a considerable increase in speed for users in the uplink, although its capacity is not as significant as HSDPA, because of the differences between those links. First of all, the majority of the data flows in the downlink direction. Secondly, in normal conditions there should be a large number of UEs communicating with the BS in the uplink, unlike downlink where there is only one transmitter per cell (the BS). So, it becomes comprehensible that providing the same performance in the forward and reverse link is not as simple as it seems, because of the restrictions imposed by the facts already discussed.

### 1.4.6 3.9G

Evolved HSPA is also known as HSPA+, HSPA Evolution and even Internet HSPA (I-HSPA). By its name it can be seen that Evolved HSPA is an enhanced version of the 3G HSPA or High Speed Packet Access system that was used to increase the speeds of the basic 3G system. Using Evolved HSPA / HSPA+ the data transfer rates are enhanced further over those that could be achieved using HSPA and other factors such as latency and the backhaul have also been addressed.

The need for HSPA+ arose out of the increasing use of data and users wanting download speeds that were comparable with fixed broadband lines. Many other applications were also starting to need much faster data transfer rates and lower levels of latency. These are addressed by the use of HSPA+.

The definition of HSPA+ / Evolved HSPA has been included in Releases 7 and 8 of the 3GPP standards. Each of these releases was equally important and described in Figure 9.

MIMO technology, which is referred to Figure 9, is used to increase the overall bitrate through transmission of two (or more) different data streams on two (or more) different antennas - using the same channelization codes at the same time, separated through use of different data precoding and different pilot channels transmitted from each Tx-antenna - to be received by two or more Rx-antennas. A simplified illustration of 2x2 MIMO is shown in Figure 8.

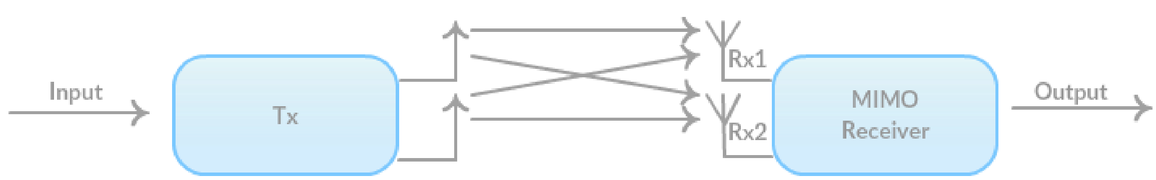
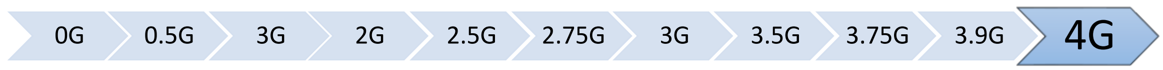



***Figure 8.*** Simplified illustration of 2x2 MIMO (Spatial Multiplexing)


***Figure 9.*** 3GPP Releases of 3.9G

## 1.5 4G "Faster & Better Mobile Broadband"

"4G LTE delivers more capacity for faster and better mobile broadband experiences, and is also expanding in to new frontiers"




### 1.5.1 Introduction

4G is a network that operates on Internet technology, combines it with other applications and technologies such as Wi-Fi and WiMAX, and runs at speeds ranging from 100Mbps to 1Gbps [12]. All these capabilities, plus some more, are defined by IMT-Advanced, which emerged as the authority framework to what constitutes 4G. So, the key features that import a system to the 4G family of standards are the following:

- high degree of commonality of functionality worldwide while retaining the flexibility to support a wide range of services and applications in a cost efficient manner;
- compatibility of services within IMT and with fixed networks;
- capability of interworking with other radio access systems;
- high-quality mobile services;
- user equipment suitable for worldwide use;
- user-friendly applications, services and equipment;
- worldwide roaming capability;
- enhanced peak data rates to support advanced services and applications (100 Mbit/s for high and 1 Gbit/s for low mobility were established as targets for research).

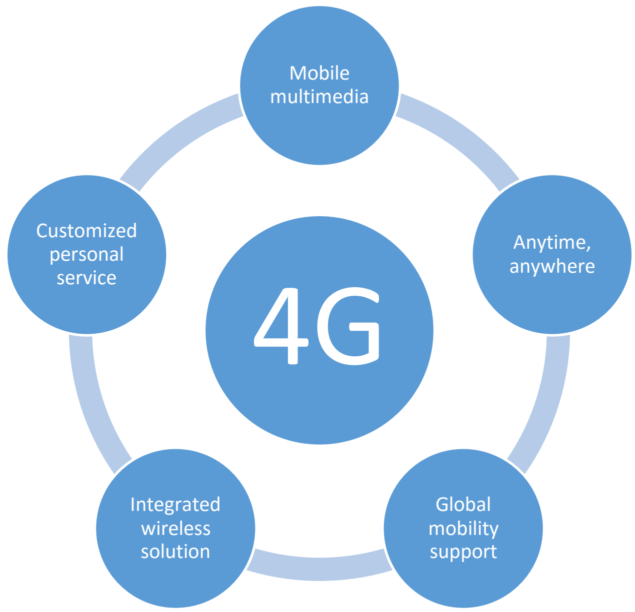



***Figure 10.*** 4G Main Features

These new features enable IMT-Advanced to address evolving user needs. Some of the services that 4G introduced to the users are:

- HD streaming
- Mobile TV
- Virtual Navigation
- Traffic Control
- Packet switched voice calls

Apart from these newly introduced applications, 4G enables services like video conferencing and telemedicine to run smoother than they did before. What becomes comprehensible, is that IMT-Advanced (4G) systems are mobile systems that that go beyond those of IMT-2000 (3G), satisfying the increasing end user demands.

### 1.5.2 4G Technologies

Following the submission and evaluation of proposals, IMT-Advanced announced in October 2010 that two systems met the requirements of IMT-Advanced. One system was LTE, while the other was an enhanced version of WiMAX under IEEE specification 802.16m, known as mobile WiMAX 2.0.

There was also a third 4G system proposed by Qualcomm, named UMB ( Ultra Mobile Broadband), that was intended to operate as a successor to cdma2000. However this project was dropped in 2008, favoring LTE, for the following reasons:

- UMB was not backwards compatible with cdma2000, in the way that cdma2000 had been with IS-95.
- Secondly, it was no longer the only system that could operate in the narrow bandwidths that dominated North America, due to the flexible bandwidth support of LTE.

### 1.5.3 Advantages & Disadvantages

Analysts use the analogy of standard vs HD TV to describe the difference between 3G and 4G.

Advantages:

- Pure data network, no circuit switched network (Figure 11).
- Higher speed, higher capacity.
- Low cost per bit
- Faster response time: One benefit of 4G technology is faster response time or lower latency. 4G technology reduces latency to 1/100th of a second (about 10ms).
- Global access, Service portability and scalable mobile services and variety of quality of services provided.
- Seamless network of multiple protocol and air interfaces.


*Figure 11.* Evolution of networks, 1G-4G

Disadvantages:

- Battery usage is more.
- Hard to implement.
- Need complicated hardware.
- The equipment required to implement a next generation network is still very expensive.
- Carriers and providers have to plan carefully to make sure that expenses are kept realistic.

# 2. Long Term Evolution

## 2.1 Introduction

LTE is a 4G mobile network technology. Mobile networks have evolved through a series of innovations to meet the ever-growing demand for wireless services, beginning with the analog cellular networks introduced almost 30 years ago.

For many years, voice calls dominated the traffic in mobile telecommunication networks. The growth of mobile data was initially slow, but in the years leading up to 2010 its use started to increase dramatically (Figure 12). Actually, in the last quarter of 2009, it was the first time that the traffic generated for mobile data was higher than voice.

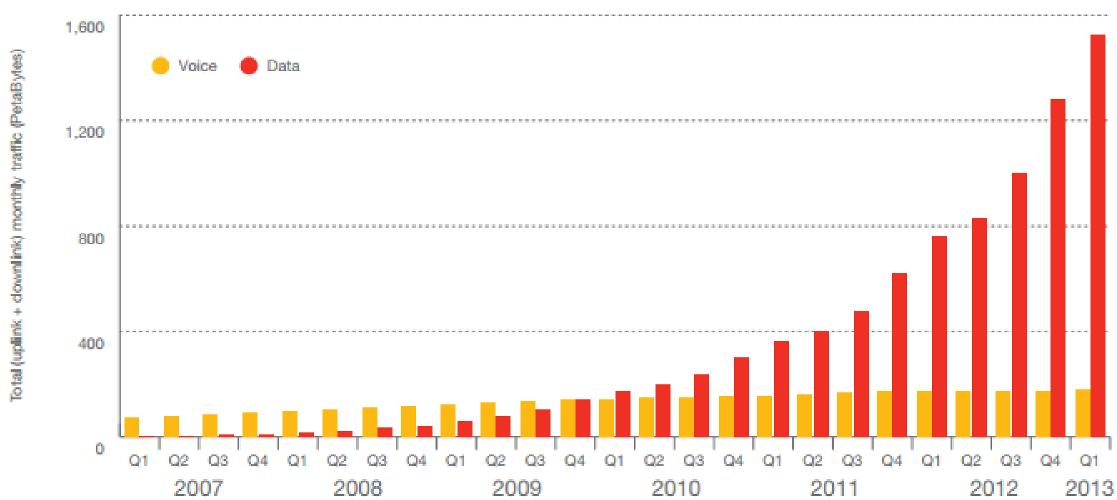



***Figure 12.*** Global total data traffic in mobile networks *[46]*

One year after, the traffic generated for mobile data was twice that for voice. Data traffic grew around 15 percent quarter-on-quarter and 55 percent year-on-year. This sudden growth in the need for data was mainly driven by the following two reasons:

- The introduction of Apple iPhone in 2007, followed by devices based on Google's Android operating system from 2008. These smartphones were more attractive and user-friendly than their predecessors and were designed to support the creation of applications by third party developers.
- The increased availability of 3.5G and 4G communications.

As a result of these issues, 2G and 3G became obsolete in the years around 2010, leading to a requirement to increase network capacity, so LTE emerged to fill this gap. Thus, the need of data is driving the move to LTE from the user's plane. As for the operators, they take advantage of LTE by the following means. Until the announcement of 4G they had to maintain 2 core networks: the circuit-switched domain for voice and the packet-switched domain for data. Provided that the network is not too congested, however, it is also possible to transport voice calls over packet switched networks using techniques such as voice over IP (VoIP). By doing this, operators can move everything to the packet switched domain and reduce both their capital and operational expenditure.

In a related issue, 3G networks introduce delays of the order of 100 milliseconds for data applications, in transferring data packets between network elements and across the air interface. This is barely acceptable for voice and causes great difficulties for more demanding applications such as real-time interactive games. Thus a second driver is the wish to reduce the end-to-end delay, or latency, in the network.

Last but not least, the specifications of UMTS and GSM have become increasingly complex over the years, due to the need to add new features to the system while maintaining backwards compatibility with earlier devices. A fresh start aids the task of the designers, by letting them improve the performance of the system without the need to support legacy devices. To conclude, all the above reasons together made LTE to see the most aggressive deployment of any mobile technology in history.

## 2.2 Evolution towards LTE

LTE is developed by 3GPP (Third Generation Partnership Project), which was established in 1998 to develop specifications for advanced mobile communications. This project has a well-established evolutionary scheme that played an important role in the formation of modern telecommunications. This evolutionary path is described briefly in the following diagram (Figure 13).

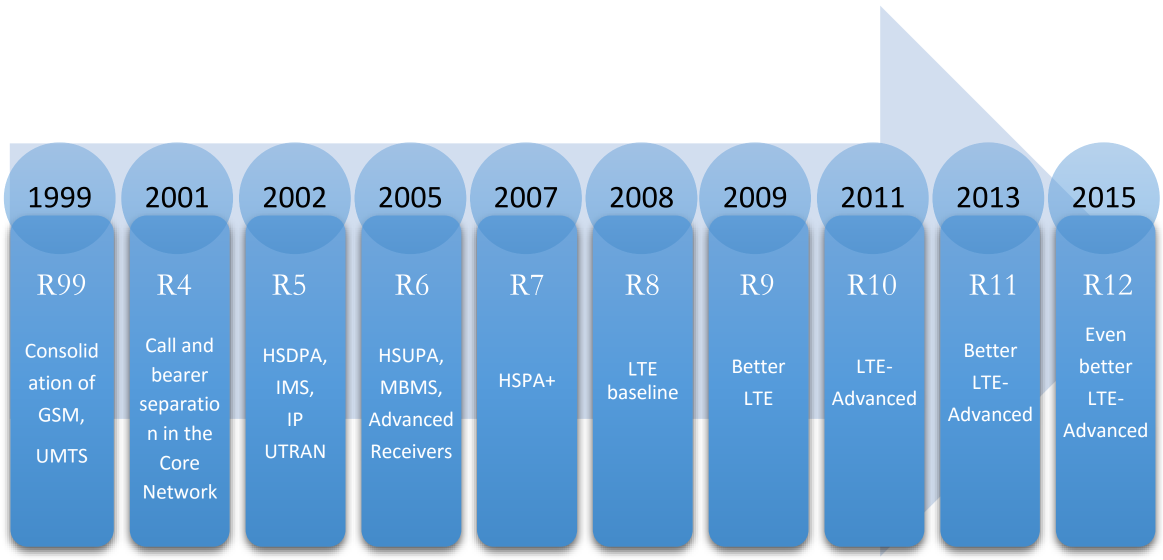



***Figure 13***. Evolutionary path of LTE towards the releases of 3GPP

The specifications produced by the Third Generation Partnership Project are organized into releases, each of which contains a stable and clearly defined set of features. The use of releases allows equipment manufacturers to build devices using some or all of the features of earlier releases, while 3GPP continues to add new features to the system in a later release. Within each release, the specifications progress through a number of different versions. New functionality can be added to successive versions until the date the release will be frozen, then the only changes involve refinement of the technical details, corrections and clarifications.

LTEs baseline was first introduced in Release 8, which was frozen in December 2008. This release contains most of the important features of LTE. In specifying Release 8, however, 3GPP omitted some of the less important features of the system. These features were eventually included in Release 9. Release 10 includes the extra capabilities that are required for LTE-Advanced. 3GPP have also continued to add new features to UMTS throughout Releases 8 to 11. This process allows network operators who stick with UMTS to remain competitive, even while other operators move over to LTE.

Perhaps the most compact upgrade among LTE and the previous generations has to do with the system architecture that is shown in Figure 14.

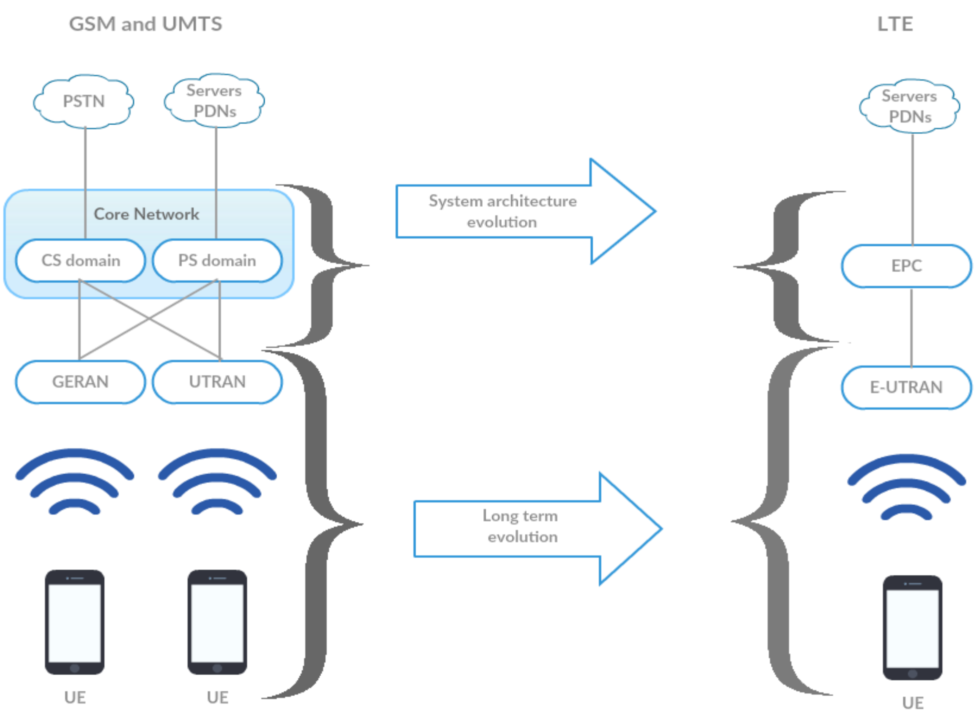



***Figure 14***. Architecture evolution

In the new architecture, the evolved packet core (EPC) is a direct replacement for the packet switched domain of UMTS and GSM. It distributes all types of information to the user, voice as well as data, using the packet switching technologies that have traditionally been used for data alone. There is no equivalent to the circuit switched domain: instead, voice calls are transported using voice over IP. The evolved UMTS terrestrial radio access network (E-UTRAN) handles the EPC's radio communications with the mobile, so is a direct replacement for the UTRAN. The mobile is still known as the user equipment, though its internal operation is very different from before.

The new architecture was designed as part of two 3GPP work items, namely system architecture evolution (SAE), which covered the core network, and long term evolution (LTE), which covered the radio access network, air interface and mobile. Officially, the whole system is known as the evolved packet system (EPS), while the acronym LTE refers only to the evolution of the air interface. Despite this official usage, LTE has become a colloquial name for the whole system, and is regularly used in this way by 3GPP.

## 2.4 LTE Air Interface Concepts

### 2.4.1 Frequency Bands

There is a growing number of LTE frequency bands that are designated as possibilities for use with LTE. Many of the LTE frequency bands are already in use for other cellular systems, whereas other LTE bands are new and being introduced as other users are re-allocated spectrum elsewhere. When UMTS was first specified in 1999, only one frequency band was defined. This band became a point around which the industry could focus its efforts in developing specifications and products. Until today, bands have been gradually added, and when LTE was specified in 2008, it inherited all the existing UMTS bands plus some new ones added in Release 8.

The frequency bands are categorized in two groups, according to the duplex method used. These are Time Division Duplexing and Frequency Division Duplexing. Thus, LTE may be deployed in two different modes, such as its predecessor (UMTS), depending on the company's preference. Although UMTS also supported these two duplex modes, LTE made the first integration steps between TDD LTE and FDD LTE, reducing the effort to support them in relation with the past.

The main differences between TDD and FDD are the following:

- Frequency Division Duplexing is implemented on a paired spectrum where downlink and uplink transmissions are sent on separate frequencies. This provides simultaneous exchange of information and reduces interference between the uplink and downlink.
- Time Division Duplexing is implemented on an unpaired spectrum, implying the usage of only one frequency for both downlink and uplink transmissions. It is suitable for asymmetric transmission demands and in cases where paired frequency is not available.

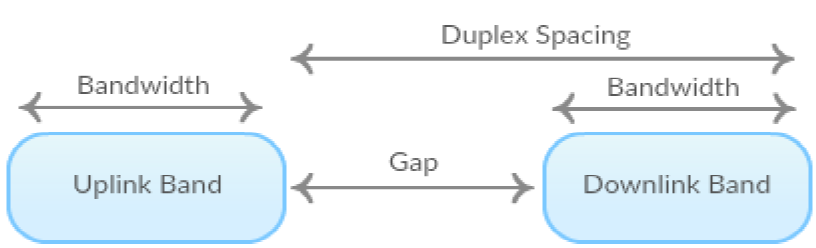



***Figure 15***. Frequency Band Terms

***Table 12.*** FDD LTE Frequency Bands

| Band Number | Uplink | | Downlink | | Bandwidth | Duplex Spacing | Gap |
|---|---|---|---|---|---|---|---|
| | Low | High | Low | High | | | |
| 1 | 1920 | 1980 | 2110 | 2170 | 60 | 190 | 130 |
| 2 | 1850 | 1910 | 1930 | 1990 | 60 | 80 | 20 |
| 3 | 1710 | 1785 | 1805 | 1880 | 75 | 95 | 20 |
| 4 | 1710 | 1755 | 2110 | 2155 | 45 | 400 | 355 |
| 5 | 824 | 849 | 869 | 894 | 25 | 45 | 20 |
| 6 | 830 | 840 | 875 | 885 | 10 | 35 | 25 |
| 7 | 2500 | 2570 | 2620 | 2690 | 70 | 120 | 50 |
| 8 | 880 | 915 | 925 | 960 | 35 | 45 | 10 |
| 9 | 1749.9 | 1784.9 | 1844.9 | 1879.9 | 35 | 95 | 60 |
| 10 | 1710 | 1770 | 2110 | 2170 | 60 | 400 | 340 |
| 11 | 1427.9 | 1452.9 | 1475.9 | 1500.9 | 20 | 48 | 23 |
| 12 | 698 | 716 | 728 | 746 | 18 | 30 | 12 |
| 13 | 777 | 787 | 746 | 756 | 10 | -31 | 21 |
| 14 | 788 | 798 | 758 | 768 | 10 | -30 | 20 |
| 15* | 1900 | 1920 | 2600 | 2620 | 20 | 700 | 680 |
| 16* | 2010 | 2025 | 2585 | 2600 | 15 | 575 | 560 |
| 17 | 704 | 716 | 734 | 746 | 12 | 30 | 18 |
| 18 | 815 | 830 | 860 | 875 | 15 | 45 | 30 |
| 19 | 830 | 845 | 875 | 890 | 15 | 45 | 30 |
| 20 | 832 | 862 | 791 | 821 | 30 | -41 | 11 |
| 21 | 1447.9 | 1462.9 | 1495.9 | 1510.9 | 15 | 48 | 33 |
| 22 | 3410 | 3490 | 3510 | 3590 | 80 | 100 | 20 |
| 23 | 2000 | 2020 | 2180 | 2200 | 20 | 180 | 160 |
| 24 | 1626.5 | 1660.5 | 1525 | 1559 | 34 | -101.5 | 67.5 |
| 25 | 1850 | 1915 | 1930 | 1995 | 65 | 80 | 15 |
| 26 | 814 | 849 | 859 | 894 | 35 | 45 | 10 |
| 27 | 807 | 824 | 852 | 869 | 17 | 45 | 28 |
| 28 | 703 | 748 | 758 | 803 | 45 | 55 | 10 |
| 29 | DL only | | 717 | 728 | 11 | 0 | 0 |
| 30 | 2305 | 2315 | 2350 | 2360 | 10 | 45 | 35 |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 31 | 452.5 | 457.5 | 462.5 | 467.5 | 5 | 10 | 5 |
| 32 | DL only | | 1452 | 1496 | 44 | 0 | 0 |
| 33 | 1920 | 2000 | 2110 | 2200 | 80 / 90 | 190 | 110/100 |
| 34 | 1710 | 1780 | 2110 | 2200 | 70 / 90 | 400 | 330/310 |
| 35 | DL only | | 5150 | 5250 | 100 | 0 | 0 |
| 36 | DL only | | 5725 | 5850 | 125 | 0 | 0 |

* These have been defined by ETSI in TS 102 735 [4] for ITU Region 1 (Europe, Middle East, and Africa) only. These bands have not been adopted at this time by ITU Region 2 (Americas) or Region 3 (Asia).

***Table 13.*** TDD LTE Frequency Bands

| Band Number | Uplink | | Downlink | | Bandwidth | Duplex Spacing | Gap |
|---|---|---|---|---|---|---|---|
| | Low | High | Low | High | | | |
| 1 | 1900 | 1920 | 1900 | 1920 | 20 | 0 | 0 |
| 2 | 2010 | 2025 | 2010 | 2025 | 15 | 0 | 0 |
| 3 | 1850 | 1910 | 1850 | 1910 | 60 | 0 | 0 |
| 4 | 1930 | 1990 | 1930 | 1990 | 60 | 0 | 0 |
| 5 | 1910 | 1930 | 1910 | 1930 | 20 | 0 | 0 |
| 6 | 2570 | 2620 | 2570 | 2620 | 50 | 0 | 0 |
| 7 | 1880 | 1920 | 1880 | 1920 | 40 | 0 | 0 |
| 8 | 2300 | 2400 | 2300 | 2400 | 100 | 0 | 0 |
| 9 | 2496 | 2690 | 2496 | 2690 | 194 | 0 | 0 |
| 10 | 3400 | 3600 | 3400 | 3600 | 200 | 0 | 0 |
| 11 | 3600 | 3800 | 3600 | 3800 | 200 | 0 | 0 |
| 12 | 703 | 803 | 703 | 803 | 100 | 0 | 0 |

Among TDD LTE an FDD LTE, time division might be the most widespread. The reasons why this happens are summarized below:

- It enables dynamic allocation of DL and UL resources to efficiently support asymmetric DL/UL traffic.
- It ensures channel reciprocity for better support of link adaptation; MIMO and other closed-loop advanced antenna techniques such as transmit beam-forming.
- Unlike FDD, which requires a pair of channels, TDD only requires a single channel for both downlink and uplink providing greater flexibility for adaptation to varied global spectrum allocations.

- Transceiver designs for TDD implementations are less complex and therefore less expensive (restrictions in the number of DL/UL switching points).

### 2.4.2 Available Channel Bandwidths

LTE is able to scale its channel bandwidth linearly without changing the underlying properties of the physical layer—these being the subcarrier spacing and the symbol length. It is sufficient to say at this point that LTE was initially designed to support six different channel bandwidths. These are 1.4 MHz, 3 MHz, 5 MHz, 10 MHz, 15 MHz, and 20 MHz. Earlier versions of the specifications also supported 1.6 MHz and 3.2 MHz for interworking with TD-SCDMA, but these were removed when the LTE TDD frame structure was aligned with the FDD frame structure rather than the TD-SCDMA frame structure from UMTS.

### 2.4.3 Reference Measurement Channels

Reference Measurement Channels exist for both the downlink and uplink and are used throughout the RF specifications to precisely describe the configuration of signals used to test the UE and BS transmitters and also their receivers. A similar principle was used in UMTS and the main difference for LTE is the use of SC-FDMA rather than W-CDMA for the air interface.

There is a plethora of reference measurement channels defined in TS 36.101 for downlink and TS 36.104 documents, so we are able to find the one that matches our needs. For example, there are RMC's available for both FDD and TDD LTE systems. The modulation is another variable the developer can specify along with the channel bandwidth.

In Table 14, the signal R.3 describes the downlink direction that is used in my thesis simulations. As we can see, TDD is configured with the modulation scheme being 16QAM. Moreover the bandwidth is set to 10 MHz leading to a fixed set of 50 resource blocks.

***Table 14.*** Reference Measurement Channel configuration parameters for downlink

| Parameter | Unit | Value |
|---|---|---|
| Reference Channel | | R.3 |
| Channel Bandwidth | MHz | 10 |
| Allocated Resource Blocks | | 50 |
| Subframe Configuration | | 1 |
| Special Subframe Configuration | | 4 |
| Modulation | | 16QAM |
| Target Coding Rate | | ½ |
| Information Bit Payload | | |
| Subframes 4,9 | Bits | 14112 |
| Subframes 1,6 | Bits | 11448 |
| Subframe 5 | Bits | n/a |
| Subframe 0 | Bits | 12960 |
| Number of PDCCH Symbols in each downlink subframe | | 2 |

Note that this peak figure represents the maximum transmitted data rate and it does not intend to indicate the performance of the downlink in real radio conditions. This peak figure is the reference used for specifying the expected performance, which will be specified relative to the maximum figures.

### 2.4.4 Orthogonal Frequency Division Multiplexing

Orthogonal frequency division multiplexing (OFDM) is the modulation scheme chosen for the LTE downlink. It is a form of digital multicarrier modulation that uses a large number of closely spaced subcarriers to carry data and control information. Each individual subcarrier is modulated at a low symbol rate with a conventional modulation format such as quadrature amplitude modulation (QAM). The combination of the many low-rate subcarriers provides overall data rates similar to conventional single-carrier modulation schemes using the same bandwidth.

The first practical implementation of an OFDM system came in 1985 when Telebit introduced the "Trailblazer" range of modems that reached speeds of 9600 bps. This highlighted one of the key advantages of OFDM: its ability to perform well through a low quality channel—in this case telephone lines—thereby outperforming existing solutions. From the early beginning, OFDM has become the technology that now delivers up to 10 Mbps over digital subscriber lines (DSL). Today, OFDM is widely used in applications from digital television and audio broadcasting to wireless networking and wired broadband Internet access. It has been adopted in the Wi-Fi arena where the standards like 802.11a, 802.11n, 802.11ac and 802.11ac (WiMAX). Furthermore, OFDM has multiple variants. While all these variants follow the basic format for OFDM, they have additional attributes or variations:

- COFDM (Coded Orthogonal frequency division multiplexing): A form of OFDM where error correction coding is incorporated into the signal.
- Flash OFDM: This is a variant of OFDM that was developed by Flarion and it is a fast hopped form of OFDM. It uses multiple tones and fast hopping to spread signals over a given spectrum band.
- OFDMA (Orthogonal frequency division multiple access): A scheme used to provide a multiple access capability for applications such as cellular telecommunications when using OFDM technologies.
- VOFDM (Vector OFDM): This form of OFDM uses the concept of MIMO technology. It is being developed by CISCO Systems. MIMO stands for Multiple Input Multiple output and it uses multiple antennas to transmit and receive the signals so that multi-path effects can be utilized to enhance the signal reception and improve the transmission speeds that can be supported.
- WOFDM (Wideband OFDM): The concept of this form of OFDM is that it uses a degree of spacing between the channels that is large enough that any frequency errors between transmitter and receiver do not affect the performance. It is particularly applicable to Wi-Fi systems.

LTE in particular uses the OFDMA (Orthogonal Frequency Division Multiple Access) variant for downlink (Figure 16).

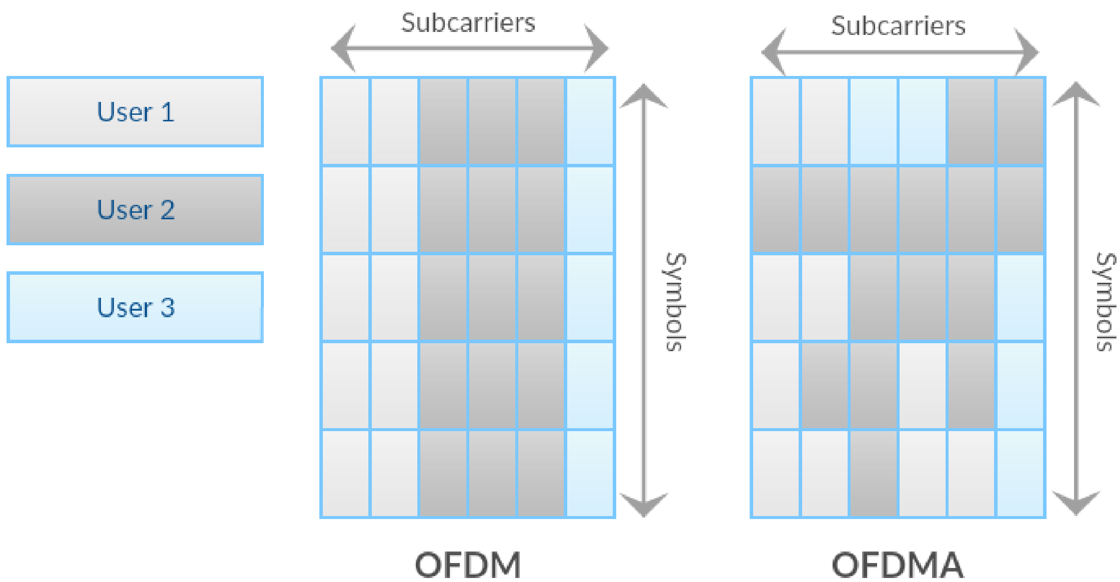



***Figure 16.*** Discrepancies between OFDM & OFDMA

The reason LTE does not use standard OFDM is that the subcarrier allocations are fixed for each user and performance can suffer from narrowband fading and interference, as depicted in Figure 16. On the other hand, OFDMA incorporates elements of time division multiple access (TDMA) so the subcarriers can be allocated dynamically among the different users of the channel. The result is a more robust system with increased capacity. The capacity comes from the trunking efficiency gained by multiplexing low rate users onto a wider channel to provide dynamic capacity when needed, and the robustness comes from the ability to schedule users by frequency to avoid narrowband interference and multipath fading.

In Figure 17, the transmitter and receiver chain for OFDMA is illustrated. It is clear, that the functions provided by the receiver are the inverse of those provided by the transmitter. Moreover, the transmitter includes the Inverse Fourier transform to translate each column of Resource Elements into an OFDMA symbol. On the receiver's side, Fourier transform is included to translate each OFDMA symbol into a column of Resource Elements. Moving on, a cyclic prefix is added at the transmitter, whilst removed at the receiver. Concluding, for the signal to be transmitted across the target coverage area, the transmitter includes a Power Amplifier to increase its strength. From the receiver's perspective, there is a Low Noise Amplifier to help improve the received signal to noise ratio.

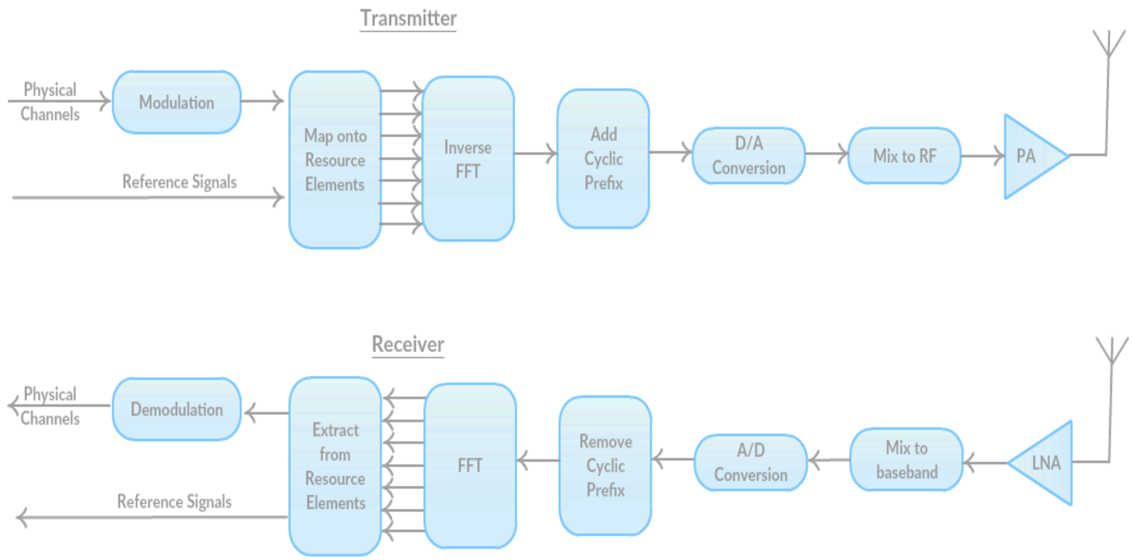



***Figure 17.*** OFDMA Transmitter and Receiver chain

### 2.4.5 Single-Carrier Frequency Division Multiple Access (SC-FDMA)

SC-FDMA was chosen for the uplink direction since it combines the low peak-to-average power ratio (PAPR) techniques of single-carrier transmission systems such as GSM and CDMA with the multipath resistance and flexible frequency allocation of OFDMA. Thus, SC-FDMA inherits all the advantages of OFDM over other well-known techniques such as TDMA and CDMA.

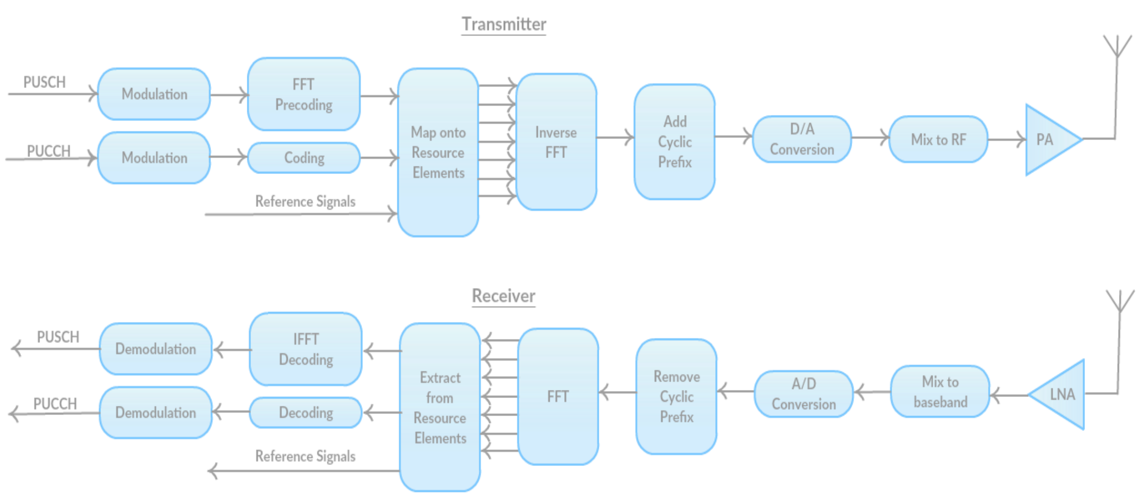



***Figure 18.*** Transmitter and Receiver chain for LTE Uplink (SC-FDMA)

As we can see in Figure 18, the transmitter in SC-FDMA combines the FFT Precoding function with the IFFT to generate the SC-FDMA symbols, while the receiver combines the FFT function with IFFT Decoding to extract the stream of modulation symbols. The above is considered as the main discrepancy among SC-FDMA & OFDMA, because in the latter the data symbols directly modulate the subcarriers. Besides that, the rest of the receiver and transmitter chain is identical.

## 2.5 LTE System Architecture Evolution

This architecture comprises the LTE evolved UMTS radio access (E-UTRA) and evolved UMTS radio access network (e-UTRAN) as well as a new evolved packet core (EPC) network. The overall network architecture is shown in Figure 19 and goes by the name enhanced packet system (EPS) or LTE/SAE; these terms are often used interchangeably. The new core network is variably called the SAE/EPC, SAE core network, or simply the EPC.

The high level requirements for LTE/SAE include:

- reduced cost per bit,
- better service provisioning,
- flexible use of new and existing frequency bands,
- simplified network architecture with open interfaces, and
- an allowance for reasonable power consumption by terminals.

To meet the requirements for LTE, LTE/SAE has been specified to achieve the following:

- Increased downlink and uplink peak data rates.
- Scalable channel bandwidths of 1.4 MHz, 3MHz, 5MHz, 10MHz, 15MHz and 20 MHz in both the uplink and the downlink.
- Spectral efficiency improvements over Release 6 HSPA of 3 to 4 times in the downlink and 2 to 3 times in the uplink.
- Sub-5ms latency for small IP packets.

- Performance optimized for low mobile speeds from 0 to 15km/h; supported with high performance from 15 to 120km/h; functional support from 120 to 350km/h. Support for 350 to 500km/h is under consideration.
- Co-existence with legacy standards while evolving toward an all-IP network.

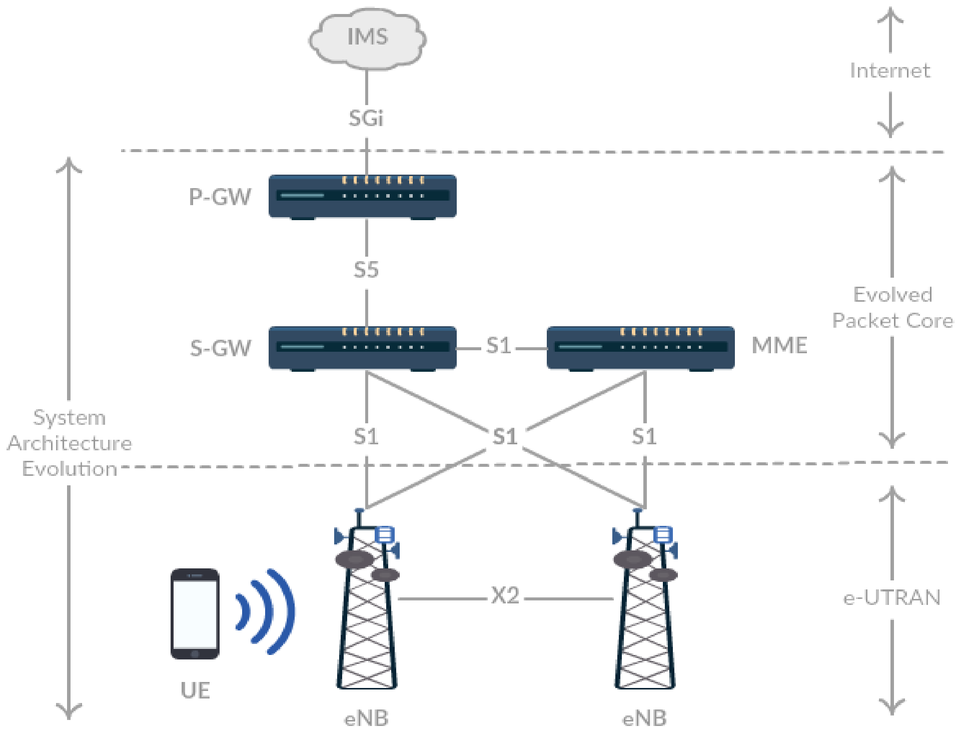



*Figure 19.* LTE SAE

## 2.5.1 e-UTRAN Architecture

The evolved UMTS terrestrial radio access network (e-UTRAN) is highlighted in Figure 20. The E-UTRAN handles the radio communications between the mobile and the evolved packet core and just has one component, the evolved Node B (eNB).

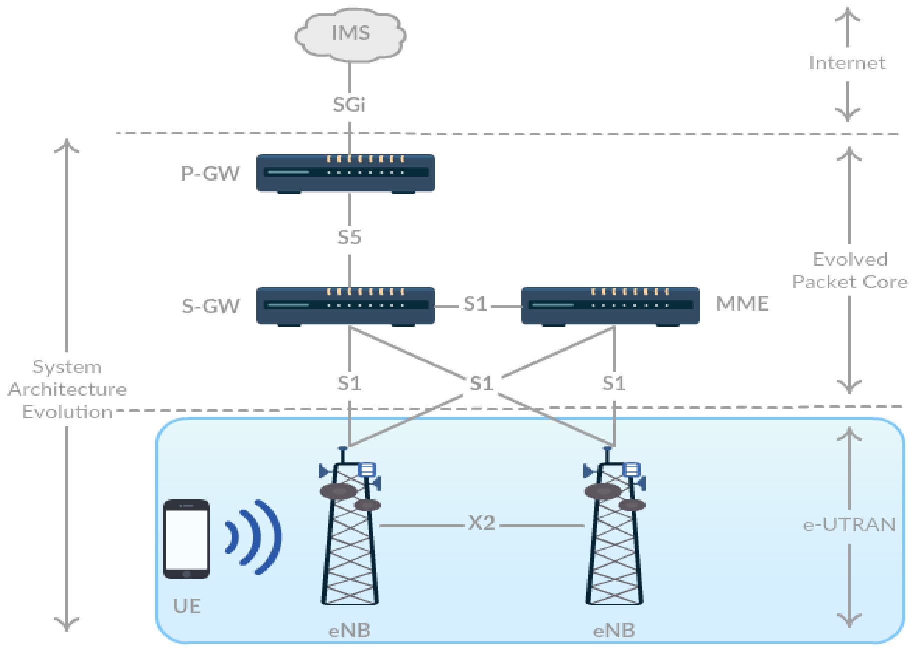



***Figure 20.*** E-UTRAN highlighted in the SAE

### 2.5.2 E-UTRAN Network Elements Explanation

Each eNB is a base station that controls the mobiles in one or more cells. A mobile communicates with just one base station and one cell at a time. The base station that is communicating with a mobile is known as its serving eNB.

The eNB has two main functions. Firstly, the eNB sends radio transmissions to all its mobiles on the downlink and receives transmissions from them on the uplink, using the analogue and digital signal processing functions of the LTE air interface. Secondly, the eNB controls the low-level operation of all its mobiles, by sending them signaling messages such as handover commands that relate to those radio transmissions. In carrying out these functions, the eNB combines the earlier functions of the Node B and the radio network controller, to reduce the latency that arises when the mobile exchanges information with the network.

### 2.5.3 Evolved Packet Core

The Evolved Packet Core (EPC) is highlighted in Figure 21.

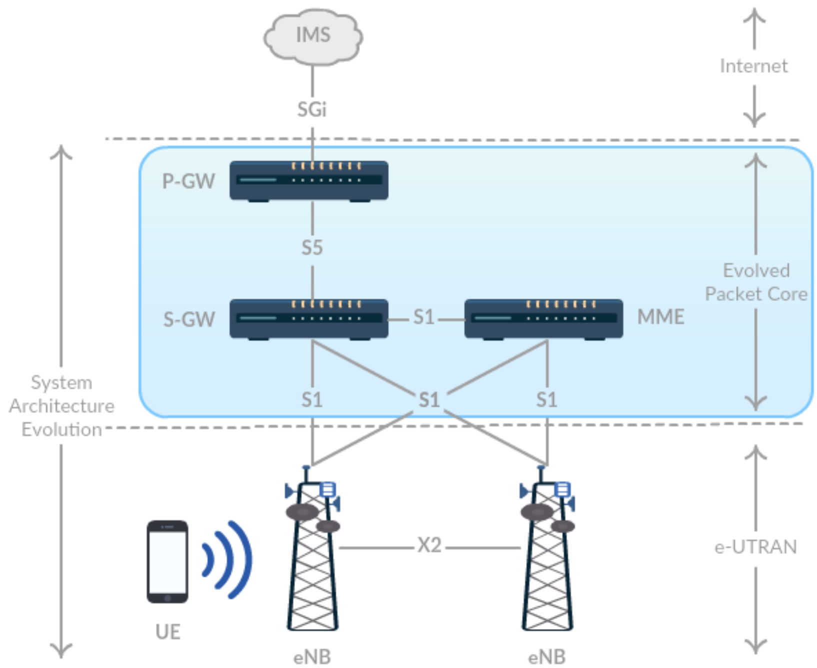



*Figure 21.* EPC highlighted in the SAE

### 2.5.4 Evolved Packet Core Elements Explanation

To begin with, each base station is connected to the EPC by means of the S1 interface. More specifically, the S1 is a point-to-point interface between an eNB within the E-UTRAN and an MME/S-GW in the EPC. Adjacent base stations are interconnected with the X2 interface, which is responsible for handover preparation and other tasks of equal importance within the E-UTRAN architecture.

The packet data network (PDN) gateway (P-GW) is the EPC's point of contact with the outside world. Through the SGi interface, each PDN gateway exchanges data with one or more external devices or packet data networks, such as the network operator's servers, the internet or the IP multimedia subsystem. The PDN GW also performs various functions such as IP address / IP prefix allocation or policy control and charging.

The serving gateway (S-GW) acts as a router, and forwards data between the base station and the PDN gateway. A typical network might contain a handful of serving gateways, each of which looks after the mobiles in a certain geographical region. Each mobile is assigned to a single serving gateway, but the serving gateway can be changed if the mobile moves sufficiently far.

Concluding, the mobility management entity (MME) controls the high-level operation of the mobile, by sending it signaling messages about issues such as security and the management of data streams that are unrelated to radio communications. As with the serving gateway, a typical network might contain a handful of MMEs, each of which looks after a certain geographical region. Each mobile is assigned to a single MME, which is known as its serving MME, but that can be changed if the mobile moves sufficiently far. The MME also controls the other elements of the network, by means of signaling messages that are internal to the EPC.

# 3. LTE Physical Layer

## 3.1 Introduction

The physical layer is responsible for the transmissions between the eNB and the UE. Existing 3G systems such as UMTS are based on CDMA. On the other hand, LTE chose a new OFDM modulation scheme, named OFDMA for the forward and SC-FDMA for the reverse direction. The reason why these modulation schemes were selected lie in the fact that there is a need for high peak transmission rate, spectral efficiency and multiple channel bandwidths.

Beyond the modulation scheme, LTE implements multiple-antenna techniques such as MIMO which can either increase channel capacity or enhance signal robustness. Together, OFDM and MIMO are two key technologies featured in LTE, differentiating its physical layer to a significant extent in relation with its predecessors.

## 3.2 Frame Structure

LTE has been defined to accommodate both paired (FDD) and unpaired spectra (TDD), so there are two types of frame structures respectively. The frame structure defines the frame, slot, and symbol in the time domain. Although the downlink and uplink utilize different multiple access schemes, they share a common frame structure.

Despite of the differences among TDD LTE and FDD LTE regarding the frame structure, they share some common features, which are the following:

- LTE frames are 10ms in duration.
- Frames are divided into 10 subframes, each of which is 1.0ms long.
- Each subframe is further divided into two slots, each of 0.5ms duration.

## 3.2.1 FDD Frame Structure

The structure of an FDD frame is shown in Figure 22.

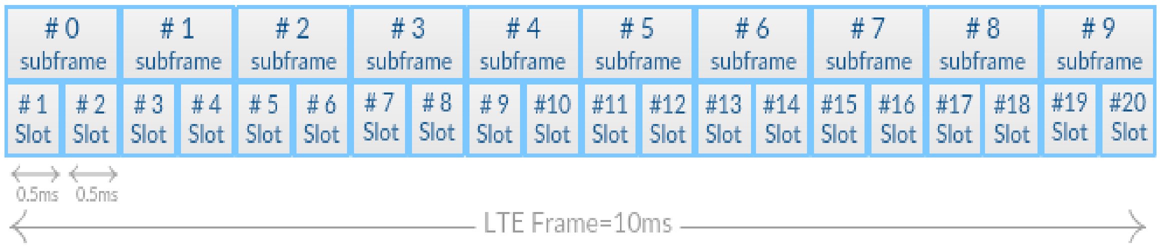



*Figure 22.* FDD Frame Structure

In FDD, both uplink and downlink share the same frame structure but use different spectra.

## 3.2.2 TDD Frame Structure

The structure of a TDD frame is shown in Figure 23.

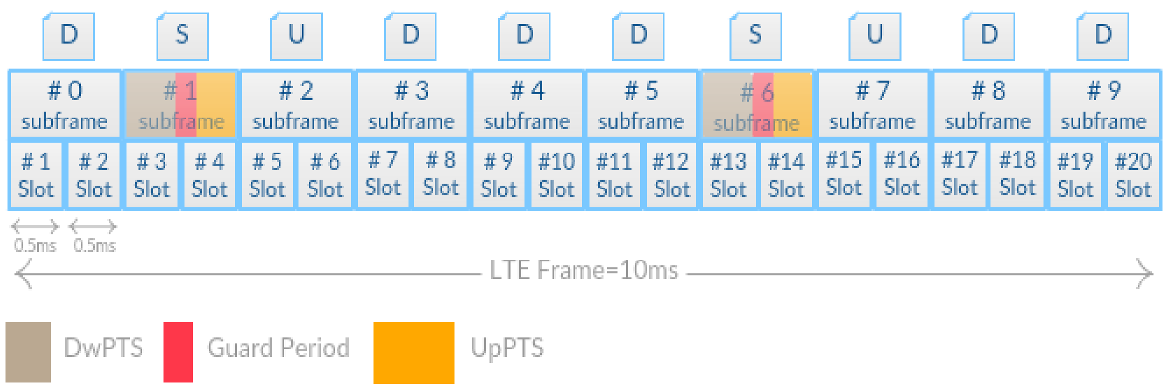



*Figure 23.* TDD Frame Structure

In TDD mode, the uplink and downlink subframes are transmitted on the same frequency and are multiplexed in the time domain. The locations of the uplink, downlink, and special subframes are determined by the frame configuration. There are seven possible configurations given in the standard, as shown in Table 15. For instance, Figure 22 illustrates a TDD frame with uplink-downlink configuration set to 2.

The special subframes that are denoted with the letter “S” in the above figure, are used for switching from downlink to uplink and contain three sections:

- DwPTS, which stands for Downlink Pilot Time Slot. DwPTS contains Primary Syncronization Signal. Physical Downlink Shared Channel can also be transmitted during DwPTS, when DwPTS is configured to be longer than a slot.
- GP is a guard period between DwPTS and UpPTS. Physical Random Access Channel format 4 begins its transmission in the guard period. Otherwise, nothing else is transmitted by this field.
- UpPTS, which stands for Uplink Pilot Time Slot. UpPTS may contain the Physical Random Access Channel and Sounding Reference Signals.

The abovementioned channels and signals that might be transmitted via DwPTS, GP and UpPTS are discussed in chapter 3.5. The lengths of DwPTS, GP and UpPTS in terms of OFDM symbols are determined by the special subframe configuration index. There are 9 possible Special Subframe configurations, as depicted in Table 16. OFDM symbols and Cyclic Prefixes mentioned in that table are further discussed in the following section.

***Table 15.*** TDD LTE Frame Configurations

| Frame Configuration | Subframe Number | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 |
| 0 | D | S | U | U | U | D | S | U | U | U |
| 1 | D | S | U | U | D | D | S | U | U | D |
| 2 | D | S | U | D | D | D | S | U | D | D |
| 3 | D | S | U | U | U | D | D | D | D | D |
| 4 | D | S | U | U | D | D | D | D | D | D |
| 5 | D | S | U | D | D | D | D | D | D | D |
| 6 | D | S | U | U | U | D | S | U | U | D |

***Table 16.*** TDD LTE Special Subframe Configurations

| Special Subframe Configuration | Normal Cyclic Prefix | | | Extended Cyclic Prefic | | |
|---|---|---|---|---|---|---|
| | DwPTS | GP | UpPTS | DwPTS | GP | UpPTS |
| 0 | 3 | 10 | 1 | 3 | 8 | 1 |
| 1 | 9 | 4 | | 8 | 3 | |
| 2 | 10 | 3 | | 9 | 2 | |
| 3 | 11 | 2 | | 10 | 1 | |
| 4 | 12 | 1 | | 3 | 7 | |
| 5 | 3 | 9 | 2 | 8 | 2 | 2 |
| 6 | 9 | 3 | | 9 | 1 | |
| 7 | 10 | 2 | | - | - | - |
| 8 | 11 | 1 | | - | - | - |

Overall, the ability to choose one out of the seven frame configurations provides to the system some sort of flexibility to change the uplink and downlink balance and characteristics. Given that, LTE is able to meet the load conditions.

## 3.3 Slot Structure

Slots consist of either 6 or 7 ODFM symbols, depending on whether the normal or extended cyclic prefix is employed.

### 3.3.1 OFDM Symbol and Cyclic Prefix

As already mentioned, each subframe contains two slots, regardless of its duplexing scheme. If we magnify the frame structure of LTE one step further, we are able to get the slot structure.

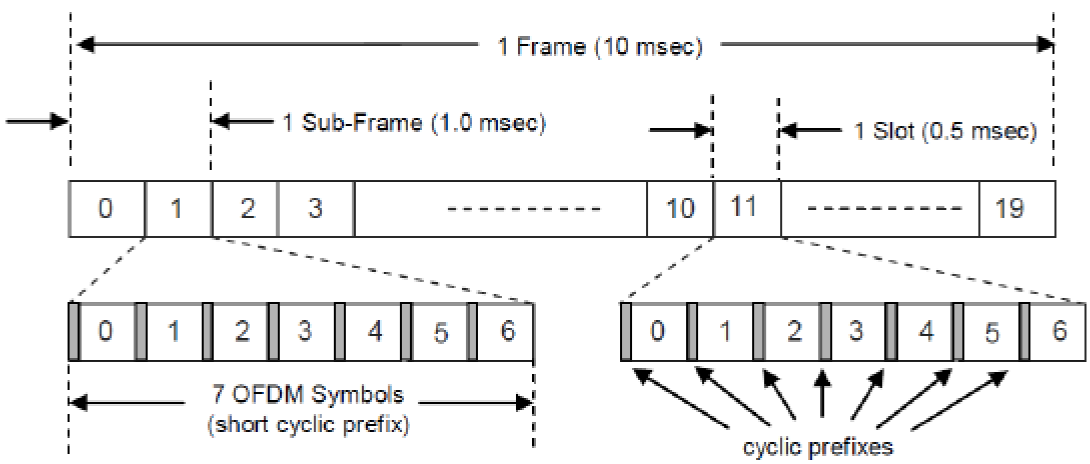



*Figure 24.* LTE Detailed Frame Structure *[13]*

As we can see in Figure 24, each subframe is divided in 7 blocks, called OFDM symbols. At the beginning of each symbol there is a Cyclic Prefix attached, while the remaining part indicates the real symbol data. There are two types of Cyclic Prefix, one is Normal and the other is Extended Cyclic Prefix which is slightly longer than the first. This augmentation impacts the slots structure, since the length of one slot is fixed and cannot be changed. Thus, if we use Extended Cyclic Prefix, the number of symbols that can be accommodated within a slot should be decreased, resulting to 6 OFDM symbols instead of 7.

The Cyclic Prefix is mandatory because it represents a guard period at the start of each OFDMA symbol, which provides protection against multi-path delay spread. The normal cyclic prefix is intended to be sufficient for the majority of scenarios, while the extended cyclic prefix is intended for scenarios with particularly high delay spread.

### 3.3.2 Resource Element and Resource Block

The basic unit in the physical layer of LTE is a resource element (RE), which spans one symbol by one subcarrier. Resource elements are grouped into resource blocks (RBs), each of which occupies 0.5 ms (one slot) in time domain and 180 kHz (12 subcarriers) in frequency domain. A resource block (RB) is the smallest unit that can be scheduled for transmission. Hence, the base station uses resource blocks for frequency dependent scheduling, by allocating the symbols and sub-carriers within each subframe in units of resource blocks. Finally, RBs are mapped into the resource grid, as illustrated in the following figure.

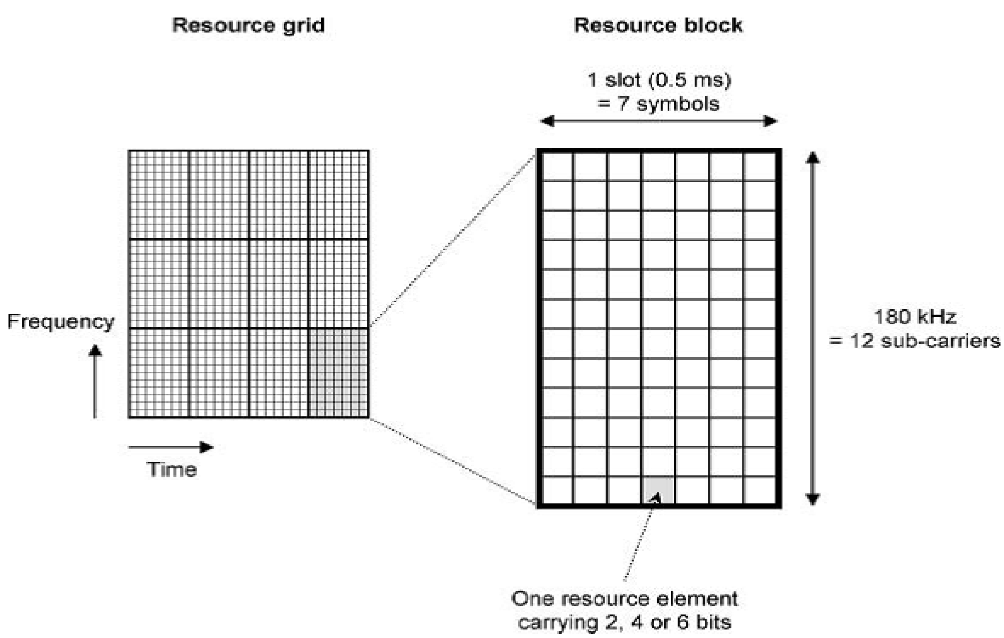



***Figure 25.*** Resource Grid *[14]*

## 3.4 Physical Signals & Channels

Physical channels convey information from higher layers in the LTE stack. This is in contrast to physical signals, which convey information that is used exclusively within the PHY layer.

### 3.4.1 Downlink Physical Signals & Channels

Table 17 defines the physical signals and channels for downlink transmission.

***Table 17.*** LTE Physical Signals & Channels

| Downlink Physical Signals | Downlink Physical Channels |
|---|---|
| Primary Synchronization Signal | Physical Downlink Shared Channel |
| Secondary Synchronization Signal | Physical Broadcast Channel |
| Cell-specific Reference Signal | Physical Downlink Control Channel |
| MBSFN Reference Signal | Physical Multicast Channel |
| UE-specific Reference Signal | Physical Control Format Indicator Channel |
| Positioning Reference Signal | Physical Hybrid Automatic Request Indicator Channel |
| Channel State Information Reference Signal | |

#### 3.4.1.1 Primary Synchronization Signal

PSS is a specific physical layer signal that is used for downlink radio frame synchronization. This signal's subcarriers are modulated using a frequency-domain Zadoff-Chu sequence. It is transmitted twice per 10ms radio frame and both transmissions are identical. The synchronization signals are always mapped to the central 62 subcarriers of the channel, which makes the cell search procedure the same, regardless of the channel bandwidth. Although 72 subcarriers (6 RB) are available, only 62 subcarriers are used so that the UE can perform the cell search procedure using an efficient length 64 fast Fourier transform.

In detail, the Primary Synchronization Signal is used to achieve subframe, slot and symbol synchronization in the time domain. Furthermore, by its default position in the frame we are able to identify the center of the channel bandwidth in the frequency domain. Last but not least, it plays an important role in the determination of the Physical Cell ID.

The exact position of PSS in the subframe is shown in Figure 26. In the case of FDD, PSS is transmitted to the last OFDM symbol of time slots 0(subframe 0) and 10(subframe 5). In TDD, it is transmitted to the third symbol of time slot 2(subframe 1) and the third symbol of time slot 12 (subframe 6). Recall that subframe 1 is always a special subframe so the PSS is sent as part of the Downlink Pilot Time Slot field, while subframe 6 may or may not be a special subframe, depending upon the uplink-downlink subframe configuration.

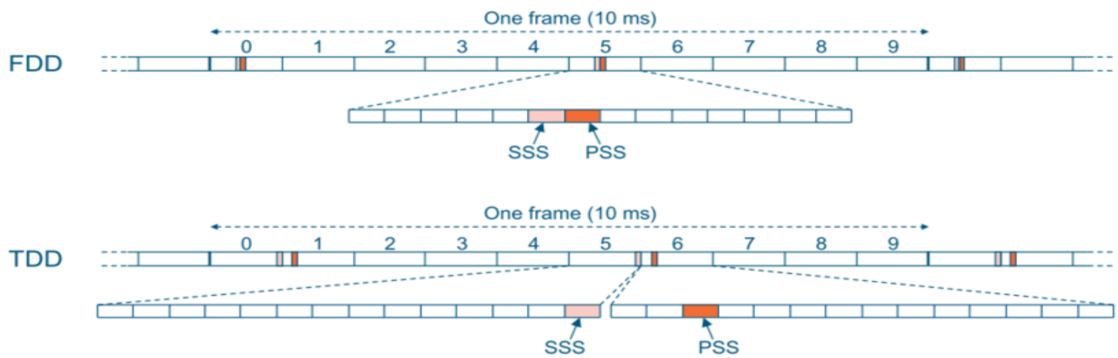



***Figure 26.*** PSS and SSS position in a LTE Frame

### 3.4.1.2 Secondary Synchronization Signal

The SSS, as part of the Synchronization Signals family has much in common with the Primary Synchronization Signal. First of all, it is also broadcasted twice within every radio frame, but in this case these two transmissions are not identical. Secondly, it is also mapped on 72 subcarriers in the middle of the band, in the exact same way as the PSS.

This signal is used primarily to achieve radio frame synchronization. Moreover, it provides important information regarding the Physical Cell ID.

The exact position of SSS is shown on Figure 26. As we can see, in the case of FDD, the SSS is located at the second last symbol of time slots 0 and 10. In TDD, it is located to the last symbol of time slot 1 (subframe 0) and the last symbol of time slot 11 (subframe 5).

### 3.4.1.3 Cell Specific Reference Signal

This type of reference signal was introduced together with the baseline of LTE, in Release 8. It is used for channel estimation and scheduling purposes.

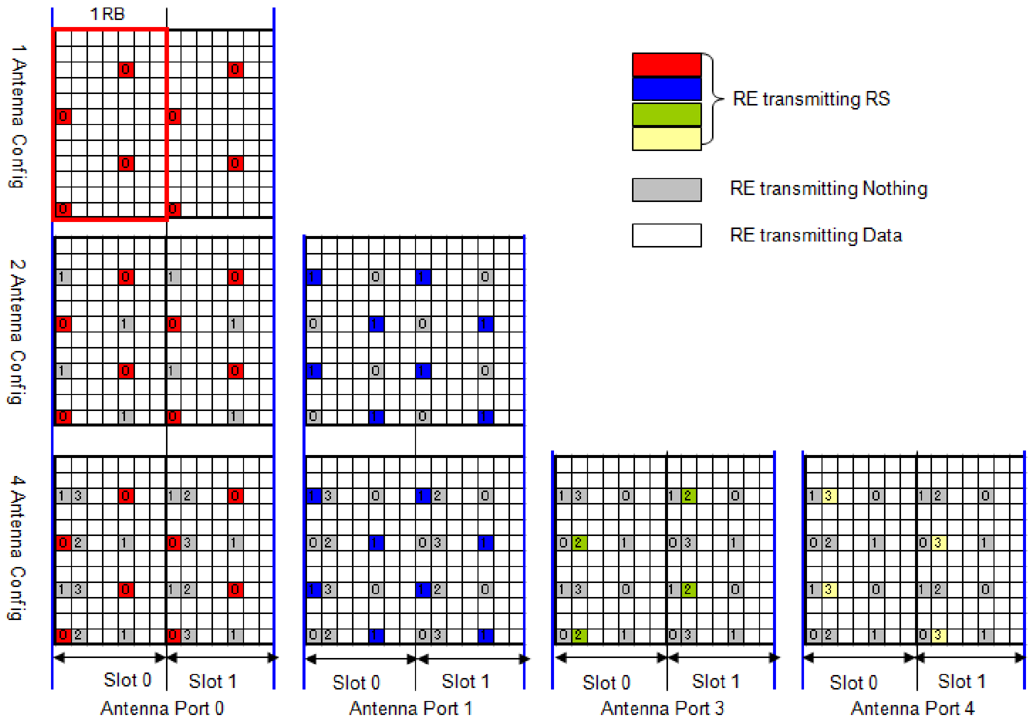



*Figure 27.* Mapping of Reference Signals in LTE downlink subframes

In detail, The CRS enables the UE to mitigate amplitude, phase, and timing errors in the received signal. The reference signals are uniformly allocated every six subcarriers in the frequency domain and every two symbols per slot in the time domain, as shown in Figure 27. From these references the UE can demodulate other downlink signals and also report the downlink channel state information on the uplink.

### 3.4.1.4 MBSFN Reference Signal

MBSFN-RS is used for equalization of PMCH transmissions. It is transmitted on antenna port 4 with PMCH and it is defined only for extended cyclic prefix. Moreover, it uses QPSK as its modulation type.

#### 3.4.1.5 UE-specific reference signal

UE-RS is the UE-specific Reference Signal and is used to enhance communication reliability by using beamforming, which is accomplished by changing the phase and magnitude of a PDSCH allocation (including UE-RS) sent to each antenna.

UE-RS was introduced in Release 8 of 3GPP, where it was transmitted on antenna port 5, within the resource blocks allocated only to a specific UE. Since Release 9, UE-RS is transmitted on antenna port 5, 7, or 8 for single-layer beamforming and on both antenna ports 7 and 8 for dual-layer beamforming, using QPSK as the modulation type.

#### 3.4.1.6 Positioning Reference Signal

This type of Reference Signals was introduced in 3GPP's Release 9. It is used to determine the location of UE based on radio access network information. One might be thinking that GPS could outshine PRS. This claim is not right, because even a minor accuracy error by the GPS could lead to further implications.

Last but not least, Positioning reference signals are allocated on virtual antenna port 6.

#### 3.4.1.7 Channel State Information Reference Signal

CSI-RS is the Channel State Information Reference Signal and is used by the UE to estimate the channel and report channel quality information (CQI) to the base station. With the addition of up to 8-layer spatial multiplexing in Release 10 came the need for 8-layer channel estimation. However, extending Cell Specific Reference Signal to 8 layers would add more signaling overhead than was desired, so the CSI Reference Signal was added.

CSI-RS is transmitted on different antenna ports (15-22) than C-RS (although likely sharing physical antennas with other antenna ports), and instead of using only time/frequency orthogonality like C-RS, CSI-RS uses code-domain orthogonality as well.

#### 3.4.1.8 Physical Broadcast Channel

PBCH carries the Master Information Block which is responsible for the correct transposition of vital information, such as the system bandwidth and the system frame number. It supports the QPSK modulation type and it is mapped on symbols 0, 1, 2, and 3 of slot 1, occupying the center 72 subcarriers.

#### 3.4.1.9 Physical Downlink Shared Channel

The Physical Downlink Shared Channel is liable for the user data transportation, on a dynamic and opportunistic basis. This channel is shared in the time domain among all the active connections and supports three modulation types, QPSK, 16QAM & 64QAM. The eNodeB is responsible for the selection of the most suitable type based on adaptation algorithm. Moreover, the radio channel conditions and buffer capacity affect the selection process.

This channel might also be used to transmit broadcast information, not transmitted on PBCH, including System Information Blocks and paging messages. Thus, a robust modulation scheme such as QPSK is considered necessary for such sensitive data.

#### 3.4.1.10 Physical Downlink Control Channel

The Physical Downlink Control Channel carries the channel allocation and control information, such as downlink resource allocation assignments and uplink grants. It occupies *n* number of OFDM symbols at the beginning of each subframe. The exact number of symbols allocated for the PDCCH is provided by the control format indicator (CFI) on Physical Control Format Indicator Channel. Acceptable CFI values are 1, 2 and 3, except in the case of a 1.4MHz channel where PDCCH might occupy the first 2, 3 or 4(CFI+1) OFDM symbols of each downlink subframe.

#### 3.4.1.11 Physical Multicast Channel

The Physical Multicast Channel contains the Multimedia Broadcast and Multicast Services (MBMS) traffic and control information. It acts in a similar way to PDSCH, but it is for reception by several terminals. It also supports QPSK, 16QAM and 64QAM modulation types.

#### 3.4.1.12 Physical Control Format Indicator Channel

The Physical Control Format Indicator Channel specifies how many OFDM symbols are used by the control channels in each downlink subframe. This value is essentially carried by the CFI (Control Format Indicator) variable, which is transmitted using the Physical Control Format Indicator Channel. As previously mentioned, this indicator is limited to the values between 1 and 3 for bandwidths greater than ten resource blocks, while in the case of 1,4 MHz bandwidth(6 resource blocks) the CFI occupies 2 to 4 OFDM symbols.

#### 3.4.1.13 Physical Hybrid ARQ Indicator Channel

The physical hybrid automatic repeat request (ARQ) indicator channel (PHICH) is the physical channel that carries the hybrid ARQ indicator (HI). The HI contains the acknowledgement/negative acknowledgement (ACK/NACK) feedback to the UE for the uplink blocks received by the eNB.

### 3.4.2 Uplink Physical Signals & Channels

Table 18 defines the physical signals and channels for uplink transmission.

***Table 18.*** LTE Uplink Physical Signals & Channels

| Uplink Physical Signals | Uplink Physical Channels |
| --- | --- |
| Demodulation Reference Signal | Physical Uplink Shared Channel |
| Sounding Reference Signal | Physical Uplink Control Channel |
| | Physical Random Access Channel |

#### 3.4.2.1 Demodulation Reference Signal

The Demodulation Reference Signal is used for synchronization and uplink channel estimation. Thus, without this signal it would be impossible for the eNodeB to decode the PUSCH.

This reference signal occupies the center symbol of each slot, meaning symbol 3 and symbol 10 of each UL subframe as shown in the figure below.

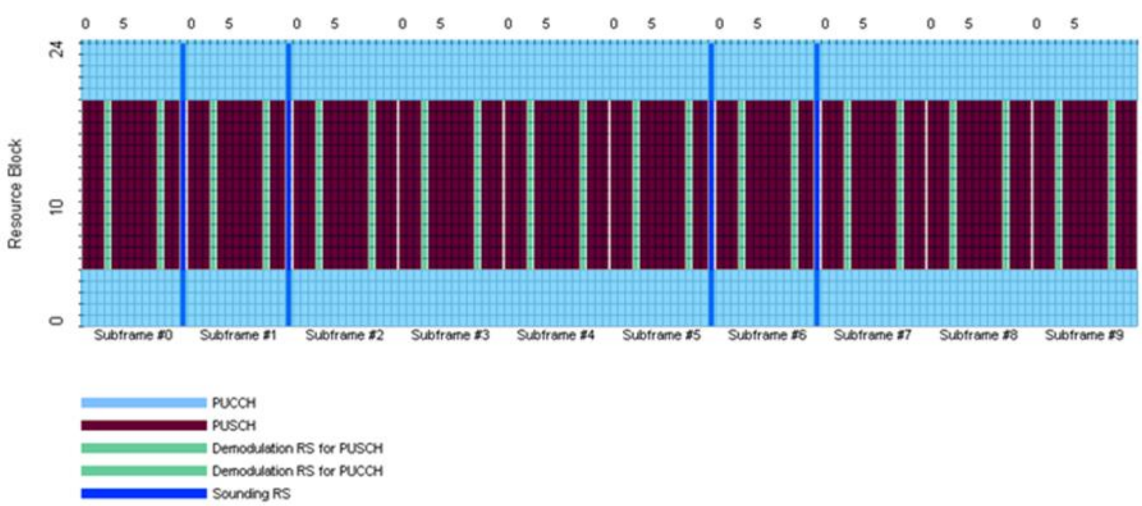



***Figure 28.*** Uplink Channels and Signals Position in a LTE frame

#### 3.4.2.2 Sounding Reference Signal

The eNB can request transmission of the SRS, which allows the eNB to estimate the uplink channel characteristics for arbitrary channel bandwidths. This estimate cannot be done using the PUCCH demodulation reference signal that is fixed to the bandwidth of the associated PUSCH/PUCCH.

The position of this reference signal in a LTE uplink frame is demonstrated in Figure 28.

#### 3.4.2.3 Physical Uplink Shared Channel (PUSCH)

The physical uplink shared channel (PUSCH) is the physical channel that carries the traffic data. It supports QPSK, 16QAM, and 64QAM modulation, although there are no performance requirements for 64QAM. Moreover, PUSCH carries the uplink shared channel (UL-SCH) transport channel and the uplink control information (UCI).

Mapping of the PUSCH is shown in Figure 28.

#### 3.4.2.4 Physical Uplink Control Channel (PUCCH)

The physical uplink control channel (PUCCH) is the physical channel that carries the uplink control information such as scheduling requests, hybrid ARQ acknowledgement /negative acknowledgement (HARQ ACK/NACK), and channel quality indicator (CQI). It is transmitted exclusively with the PUSCH from the same UE. The PUCCH supports BPSK and QPSK modulation schemes.

The position of PUCCH in a uplink frame is shown in Figure 28.

#### 3.4.2.5 Physical Random Access Channel

This channel is used to carry random access preambles used for initiation of random access procedure. This procedure can be performed to achieve uplink synchronization between UE and eNB or obtain the resource for RRC Connection Request.

### 3.4.3 Downlink Control Information Coding

Resource allocation in Long Term Evolution is impressive due to the flexibility that it provides. One of the main factors that contribute to this procedure is the DCI's (Downlink Control Indicator), which reports -among others- to the receiving side the exact location of the data and the modulation scheme used. Thus, each user knows which resource block carries their data and what kind of demodulation scheme you have to use to decode the data.

These indicators are mapped to the PDCCH in the physical layer and carry information regarding the following:

- Transport format information:
    - Modulation scheme
    - Coding Scheme
    - Redundancy version
    - New data indicator
    - Cyclic shift for demodulation RS
    - UL index
    - CQI request downlink assignment index
    - HARQ process number
    - Code word information
- Resource Allocation Information:
    - RB assignment
    - Hopping resource allocation
    - Localized/distributed virtual resource block assignment flag
    - HARQ information
- Transmit Power Control (TPC) command

Specifically, there are several different DCI formats, each of which is represented below, along with the information set it can carry:

- DCI Format 0: Used for scheduling of PUSCH in one UL cell. Carrier Indicator, Format 0/1A Flag, Frequency Hopping Flag, RB Assignment and Hopping Resource Allocation, MCS Index, NDI, TPC Command for scheduled PUSCH, Cyclic Shift for DMRS and OCC Index, UL Index (TDD only), DAI (TDD only), CQI Request, SRS Request, Multi-cluster Flag.
- DCI Format 1: Used for scheduling of one PDSCH codeword in one cell. Carrier Indicator, Resource Allocation Header, Resource Block Assignment, MCS Index,

HARQ Process Number, NDI, RV Index, TPC Command for PUCCH, DAI (TDD only).

- DCI Format 1A: Used for compact scheduling of one PDSCH codeword in one cell. Carrier Indicator, Flag for Format 0/1A, Localized/Distributed VRB Assignment Flag, Resource Block Assignment, Preamble Index, PRACH Mask Index. Used for random access procedure initiated by a PDCCH order. Flag for Format 0/1A, Localized/Distributed VRB Assignment Flag, Resource Block Assignment, MCS Index, HARQ Process Number, NDI, RV Index, TPC Command for PUCCH (RA-RNTI/P-RNTI/SI-RNTI; N_1A_PRB), DAI (TDD only).
- DCI Format 1B: Used for compact scheduling of one PDSCH codeword in one cell with precoding information. Carrier Indicator, Localized/Distributed VRB Assignment Flag, Resource Block Assignment, MCS Index, HARQ Process Number, NDI, RV Index, TPC Command for PUCCH, DAI (TDD only), Transmitted PMI (TPMI) Information for Precoding, PMI Confirmation for Precoding.
- DCI Format 1C: Used for very compact scheduling of one PDSCH codeword. Gap Indicator, Resource Block Assignment, MCS. Used for notifying MCCH change. Information for MCCH Change Notification.
- DCI Format 1D: Used for compact scheduling of one PDSCH codeword in one cell with precoding and power offset information. Carrier Indicator, Localized/Distributed VRB Assignment Flag, Resource Block Assignment, MCS Index, HARQ Process Number, NDI, RV Index, TPC Command for PUCCH, Downlink Assignment Index (TDD only), TPMI Information for Precoding, Downlink Power Offset.
- DCI Format 2: Carrier Indicator, Resource Allocation Header, Resource Block Assignment, TPC Command for PUCCH, Downlink Assignment Index (TDD only), HARQ Process Number, Transport Block to Codeword Swap Flag, MCS Index/NDI/RV Index for Transport Block 1, MCS Index/NDI/RV Index for Transport Block 2, Precoding Information.
- DCI Format 2A: Carrier Indicator, Resource Allocation Header, Resource Block Assignment, TPC Command for PUCCH, Downlink Assignment Index (TDD only), HARQ Process Number, Transport Block to Codeword Swap Flag, MCS

Index/NDI/RV Index for Transport Block 1, MCS Index/NDI/RV Index for Transport Block 2, Precoding Information.

- DCI Format 2B: Carrier Indicator, Resource Allocation Header, Resource Block Assignment, TPC Command for PUCCH, Downlink Assignment Index (TDD only), HARQ Process Number, Scrambling Identity, Transport Block to Codeword Swap Flag, MCS Index/NDI/RV Index for Transport Block 1, MCS Index/NDI/RV Index for Transport Block 2
- DCI Format 2C: Carrier Indicator, Resource Allocation Header, Resource Block Assignment, TPC Command for PUCCH, Downlink Assignment Index (TDD only), HARQ Process Number, Antenna Port/Scrambling Identity/Number of Layers, Transport Block to Codeword Swap Flag, MCS Index/NDI/RV Index for Transport Block 1, MCS Index/NDI/RV Index for Transport Block 2.
- DCI Format 3: Used for the transmission of TPC commands for PUCCH and PUSCH with 2-bit power adjustments. 2-bits TPC Command for PUCCH/PUSCH (TPC #1, #2, ...).
- DCI Format 3A: Used for the transmission of TPC commands for PUCCH and PUSCH with single bit power adjustments. 1-bit TPC Command for PUCCH/PUSCH (TPC #1, #2, ...).
- DCI Format 4: Used for the scheduling of PUSCH in one UL cell with multi-antenna port transmission mode. Carrier Indicator, Resource Block Assignment and Hopping Resource Allocation, TPC Command for scheduled PUSCH, Cyclic Shift for DMRS and OCC Index, UL Index (TDD only), DAI (TDD only), CQI Request, SRS Request, Multi-cluster Flag, MCS/NDI for Transport Blcok1, MCS/NDI for Transport Block2, Precoding Information and Number of Layer.

In the following figure we are able to see an illustration on how the DCI performs in a typical frame.

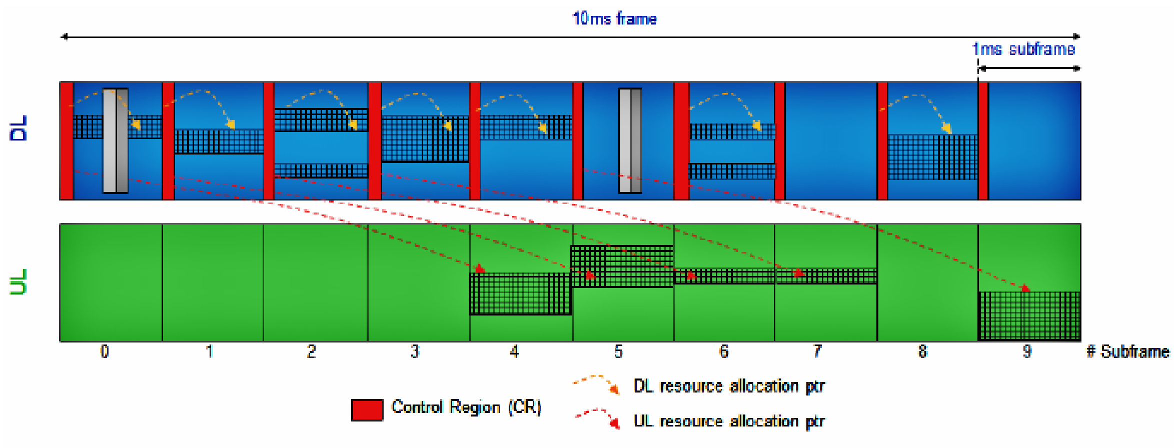



***Figure 29.*** LTE Scheduling Assignments and Grants *[28]*

# 4. Advanced Scheduling Algorithms

Scheduling is a very important method of resource assigning by base station to users in LTE (Long Term Evolution) systems. It is in charge of distributing the available resources among active users in order to satisfy their QoS needs.

Scheduling is a simple task if there is only one user and if there is less data waiting in the transmission buffer than can be sent over the air interface. When the eNode-B serves several users, or several bearers to be precise, and the amount of data in the downlink buffer exceeds that which can be sent in a subframe, then the scheduler has to decide which users and bearers are given an assignment grant for the next subframe and how much capacity is allocated to each. This decision is influenced by several factors.

If a certain bandwidth, delay and jitter have been granted for a bearer to a particular user then the scheduler has to ensure that this is met. The data of this bearer is then given preference over the data arriving from the network for other bearers for the same or a different user. In practice, however, such QoS attributes are not widely used and hence most bearers have the same priority on the radio interface.

For bearers with the same priority, other factors may influence the scheduler's decision such as when to schedule a user and how many RBs are allocated to him in each subframe. If each bearer of the same priority was treated equally, some capacity on the air interface would be wasted. With this approach, mobile devices that currently or permanently experience bad radio conditions, for example, at the cell edge, would have to be assigned a disproportional number of RBs because of the low modulation and coding scheme required. The other extreme is to always prefer users that experience very good radio conditions as this would lead to very low data rates for users experiencing bad radio conditions. As a consequence, proportional fair schedulers take the overall radio conditions into account, observe changes for each user over time and try to find a balance between the best use of the cell's overall capacity and the throughput for each user.

## 4.1 Dynamic Scheduling

In DS, queued packets of the users are scheduled every TTI by allocating the required PRBs to the users. It is essential, that each packet in the downlink direction is scheduled on PDSCH, whilst in the uplink direction the PUSCH is utilized. The allocation information for each packet is carried as DCI and specifies its exact location in PDSCH & PUSCH. Thus, the PDCCH carries all allocation information of the data that is stored in the forward and reverse link, as illustrated in the figure below.

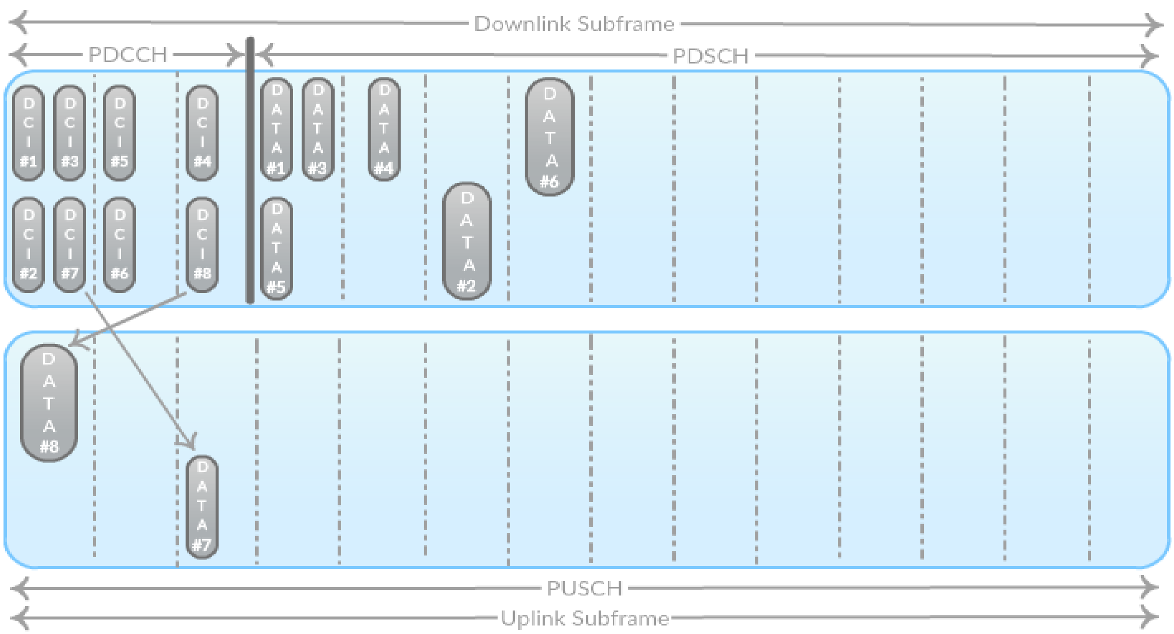



*Figure 30.* PDCCH allocation using DS

The advantage of Dynamic Scheduling is basically the flexibility to alter the size of data in each subframe. You can push more data in one SF, less on another, making it suitable for bursty, unpredictable and download type of traffic.

## 4.2 Persistent Scheduling

While dynamic scheduling is great for certain applications, on real time streaming services it is not efficient because PDCCH is limited size. Since the PDCCH is limited size, there is a limit as to how many DCIs can be carried in a subframe. This can in-turn limit the number of UEs which can receive an allocation for that subframe when using dynamic scheduling. The aforementioned scenario emerges when the packets have low data rate and arrive within short intervals, because the overhead of each packet is very high, as only little data is sent for each scheduling message.

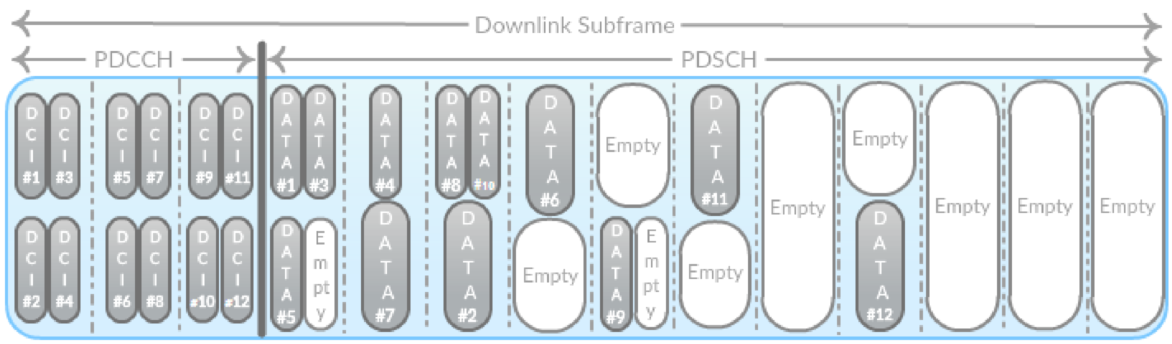



*Figure 31.* DS inefficiency scenario

Figure 31 illustrates a DS inefficiency scenario where PDCCH is not able to grant more user data due to lack of resources, whilst PDSCH is not fully utilized. Thus, it is obvious that this scheduling technique results to an unacceptable resource waste.

A type of traffic that may cause that PDCCH overflow is VoLTE, which generates approximately 300 bits of coded speech every 20ms. In such cases, the minimization of granting overhead plays a vital role in the effective utilization of bandwidth. Thereby, Persistent scheduling algorithm was introduced.

Persistent Scheduling algorithm takes advantage of the fairly consistent and predictable transmission pattern of VoLTE packets to make a persistent grant of uplink and downlink RBs rather than scheduling each uplink and download RB individually. A persistent grant removes the need to have a separate DCI for each 20ms of encoded audio. Consequently, PS reduces the processing overhead and provides more bandwidth to accommodate additional users.

Certain things remain fixed for the users that have Persistent Scheduling algorithm enabled:

- The periodicity that a PS occasion reoccurs. The UE is able to adjust the periodicity according to its needs. For example, VoLTE generates a new packet every 20ms, hence a periodicity of 20ms is used.
- The reserved resource block position. During PS a persistent resource block allocation is maintained. This way, the users/eNodeB know the exact location of their data in PDSCH/PUSCH respectively, taking away the need to overload PDCCH with a DCI for their transmissions.
- The modulation & coding scheme.

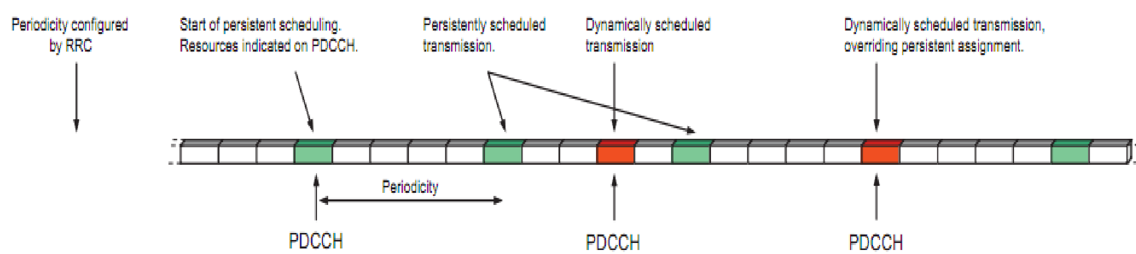



***Figure 32.*** Persistent Scheduling implementation *[35]*

To witness the advantages of PS, let's take a look at the following examples, showing how PS works in both downlink and uplink:

- Persistent scheduling algorithm is activated in DL for the users who were about to receive data packets #1 and #4 and the next DL subframe is considered an SPS occasion for these users (SPS occasion every 20ms) according to the preconfigured periodicity. The resource allocation of that DL subframe would look like the one in Figure 32.

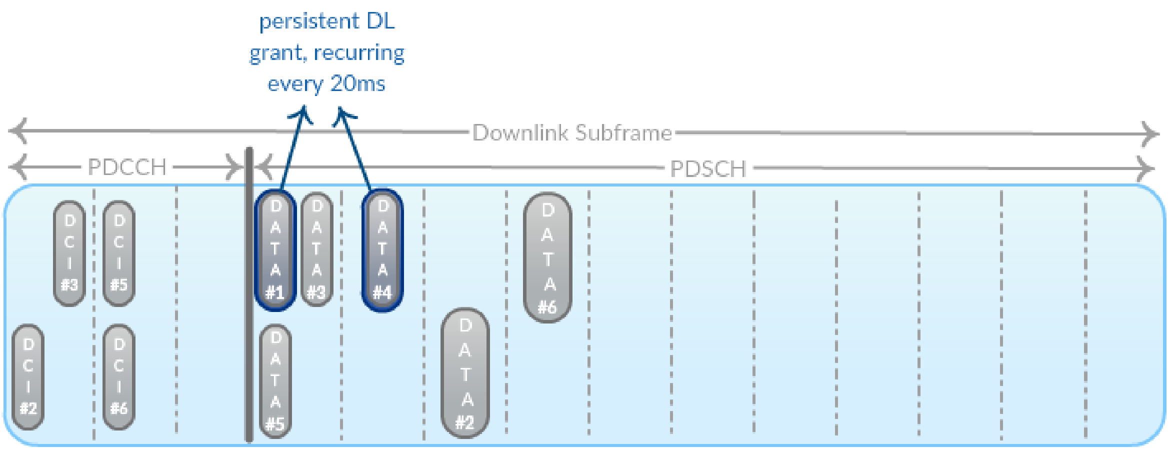



*Figure 33.* PS operation occasion in DL


As we can see in the above figure, there is no DCI for the data packets that receive a persistent DL grant. That is because we reserve a fixed RB in the subframe that comes every 20ms, due to the fact that the users who have PS enabled expect a new packet to arrive to their buffer approximately every 20ms. Thus, 20ms after the above suframe the reserved RB will still expect to be utilized by the next packet that is about to arrive, for each of the users having PS enabled. The users that are going to receive the data packets #1 and #4 do not need to decode a DCI because they already know where to find their data in the subframe, as the reserved RB has a fixed position. The aforementioned case is illustrated in the figure below.

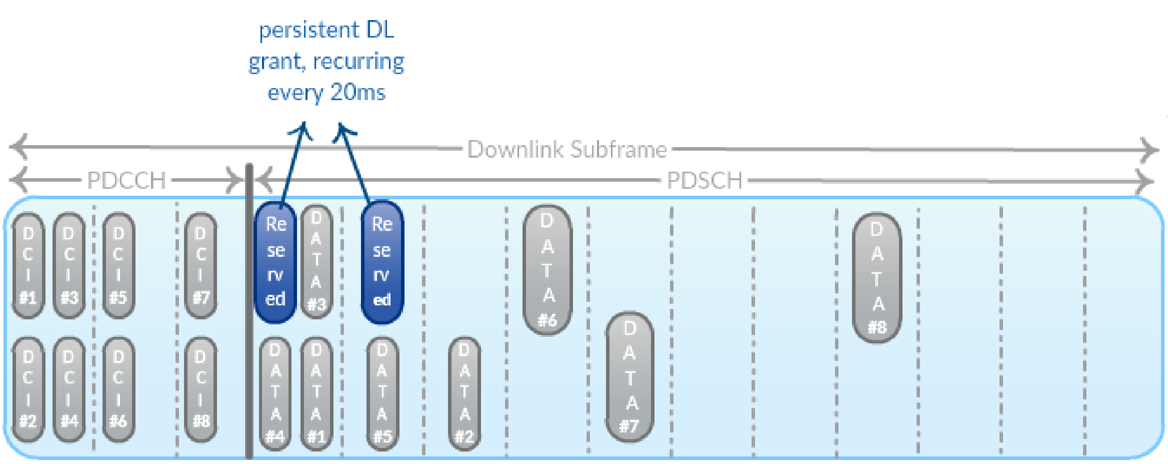



*Figure 34.* PS operation occasion in DL (after 20s)

- Persistent scheduling algorithm is enabled in UL for the users who are about to transmit data packets #7 and #8, and the next UL subframe is considered an SPS occasion (SPS occasion every 20ms) for these users according to the preconfigured periodicity, the resource allocation of that UL subframe would look like the one in Figure 35.

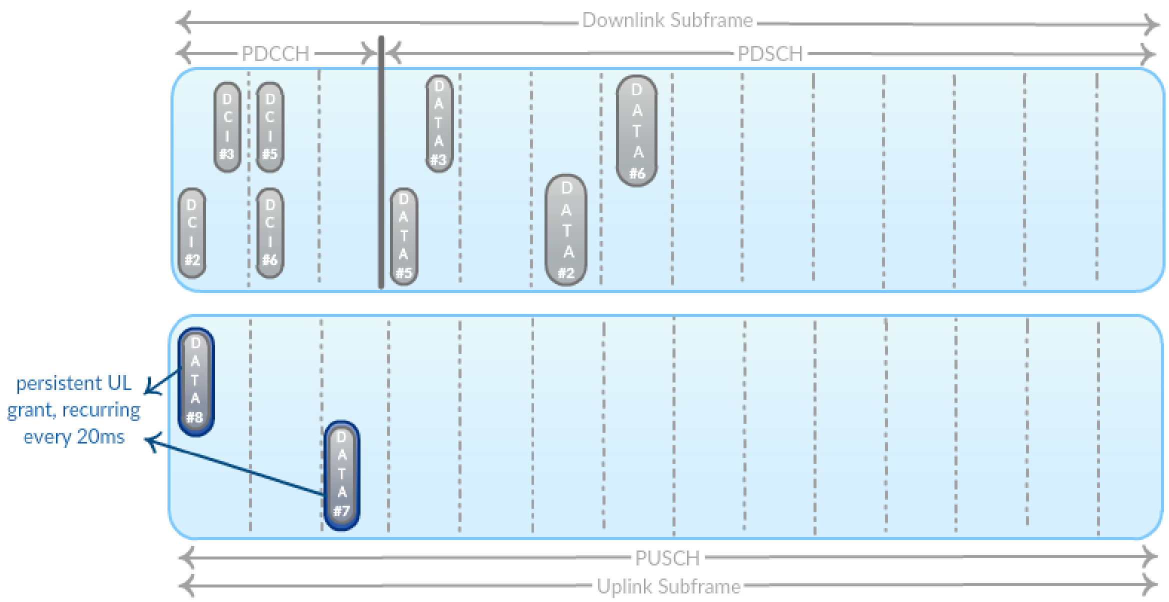



***Figure 35.*** PS operation in UL

According to the principles of Dynamic Scheduling, the PDCCH carries all allocation information of the data that is stored in the forward and reverse link. Given that, it is clear that each and every packet scheduled in the Uplink direction has to be granted by a precedent downlink subframe, of the same frame. The aforementioned precedent downlink subframe is the one shown in Figure 36. As we can see, there are no DCIs in the downlink subframe referring to the data packets #7 and #8. This happens because PS is enabled for the users that are about to transmit these two data packets, which means that all the uplink subframes recurring every 20ms will have a fixed positioned reserved RB for the expected traffic. Thus, the uplink subframe that will recur after 20ms will be like the following.

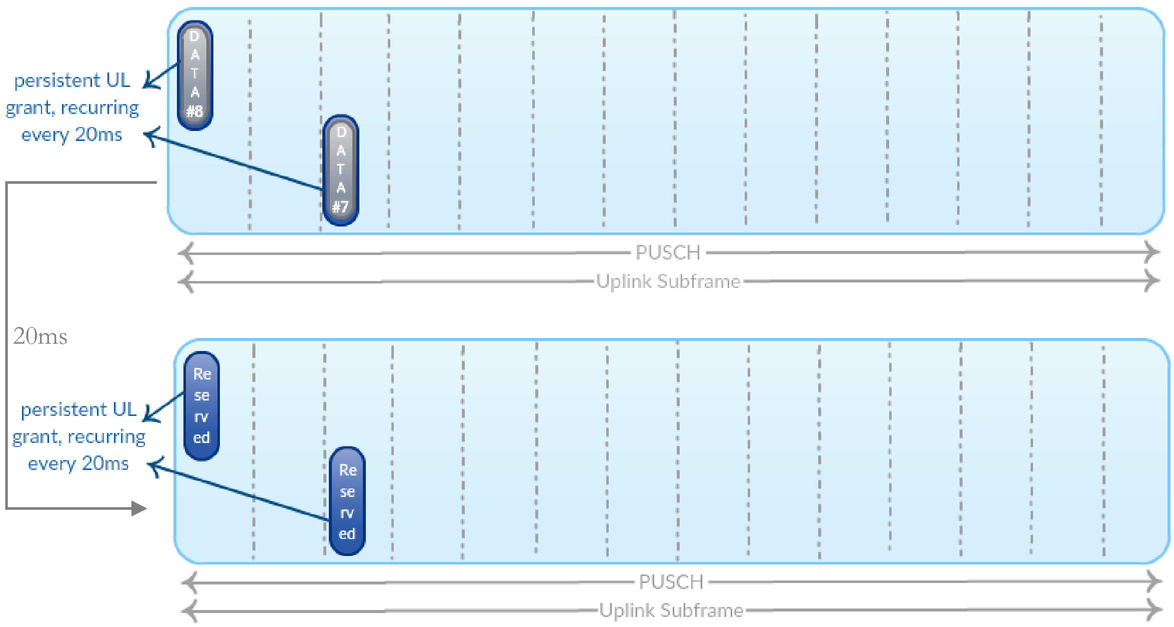



*Figure 36.* PS operation in UL

## 4.3 Semi-Persistent Scheduling

Although Persistent Scheduling seems efficient while the reserved resources are used, there is a potential downside that occurs when there is silence during a VoLTE conversation. During the silence periods the UE does not transmit audio packets, therefore the reserved resources are wasted. Moreover, the user might be walking as he holds a VoLTE conversation. That means that the radio link conditions are constantly changing, which implies to the fact that the modulation and coding scheme has to change accordingly.

In the effort to address the above inabilities, the Semi-Persistent Scheduling algorithm aroused, providing the capabilities shown in Figure 37 and 38 according to TS 36.321.

```
SPS-ConfigDL ::=     CHOICE{
    release                         NULL,
    setup                           SEQUENCE {
        semiPersistSchedIntervalDL          ENUMERATED {
                                                sf10, sf20, sf32, sf40, sf64, sf80,
                                                sf128, sf160, sf320, sf640, spare6,
                                                spare5, spare4, spare3, spare2,
                                                spare1},
        numberOfConfSPS-Processes           INTEGER (1..8),
        n1PUCCH-AN-PersistentList           N1PUCCH-AN-PersistentList,
```

*Figure 37.* SPS configuration in DL

```
SPS-ConfigUL ::=    CHOICE {
    release                     NULL,
    setup                       SEQUENCE {
        semiPersistSchedIntervalUL      ENUMERATED {
                                            sf10, sf20, sf32, sf40, sf64, sf80,
                                            sf128, sf160, sf320, sf640, spare6,
                                            spare5, spare4, spare3, spare2,
                                            spare1},
        implicitReleaseAfter            ENUMERATED {e2, e3, e4, e8},
        p0-Persistent                   SEQUENCE {
            p0-NominalPUSCH-Persistent      INTEGER (-126..24),
            p0-UE-PUSCH-Persistent          INTEGER (-8..7)
        }       OPTIONAL,                                   -- Need OP
```

***Figure 38.*** SPS configuration in UL

The first parameters illustrated in Figure 37 and Figure 38 are `semiPersistSchedIntervalDL` and `semiPersistSchedIntervalUL` that represent the periodicity within which the transmission grant shall reoccur in DL & UL respectively. This parameter is measured in subframes, meaning that value sf10 corresponds to 10 subframes, value sf20 corresponds to 20 subframes etc. For TDD-LTE, this parameter should be rounded to the nearest integer (of 10 subframes) towards zero, e.g. sf10 corresponds to 10 subframes, sf32 corresponds to 30 subframes, sf128 corresponds to 130 subframes. As an example, VoLTE uses `semiPersistSchedIntervalUL=sf20`, because the packet generation rate is approximately 20ms.

Moving on, SPS in the uplink direction implicitly releases the allocated traffic channel when a certain number of empty transmission slots are found on it. This parameter is defined in Figure 37 as `implicitReleaseAfter` and its purpose of existence is to detect and confirm an ensuing silence period, or help to hold the reserved traffic channel to prevent it from being released due to some packets being lost over the radio channel. The domain of this variable is set to {2,3,4,8}, referring to the number of consecutive new MAC PDUs each containing zero MAC SDUs after which UE will release the SPS resources.

Besides the implicit release of resources, the eNodeB can explicitly send DCI Format 0 to indicate SPS release. Upon receiving DCI Format 0 which indicates SPS release, the UE should clear the configured uplink grant.

# 5. Proposed Scheme

The proposed scheme extends the capabilities of Semi-Persistent algorithm introducing the concept of prediction in the silence period detection process. Each users talk activity pattern in a phone call is random. For example, a user might be talking for the first 5 seconds in a phone call and then remain silent for the remaining call duration, while some other user might have the talk activity pattern that is illustrated in Figure 39.

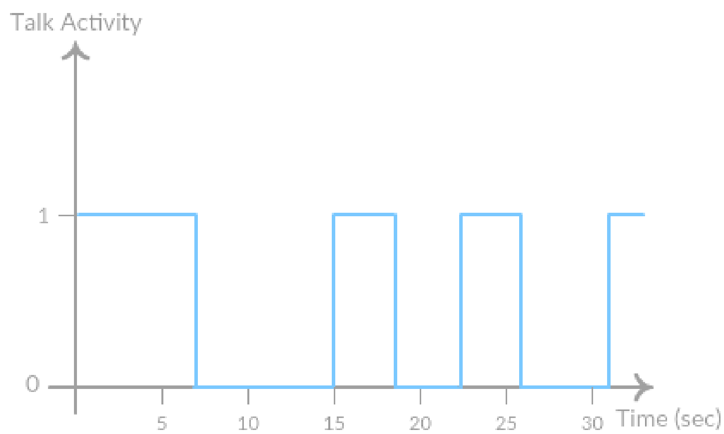



***Figure 39.*** Users Random Talk Activity Pattern during a phone call

It is clear, that each user in every phone call has a unique talk activity pattern. Regardless of that, the Semi-Persistent scheduling algorithm has to meet the QoS needs of each and every user. Thus, predicting a user's upcoming silence period directly after a talk-spurt, allows the algorithm to release the reserved resources on time and accommodate another terminal, maximizing the effective utilization of bandwidth.

The variable that specifies when the resources that are allocated to each user performing a VoLTE call will be released, is `implicitReleaseAfter`. The aim of this parameter is to adjust its value in a way that predicts as soon as possible a silence period, whilst avoid the release of resources due to a false prediction. The reasons that might lead to a false prediction are packet loss and the fluctuation observed in real world packet generation scenarios.

Given that the available values for this variable are `{2,3,4,8}` and the periodicity of Semi-Persistent scheduling is set to 20ms for a VoLTE call (Semi-Persistent occasion for a user every 20ms), if:

- `implicitReleaseAfter = 2`, after 2 empty SPS occasions (`2 * 20ms=40ms`) the reserved resources for this user will be released.
- `implicitReleaseAfter = 3`, after 3 empty SPS occasions (`3 * 20ms=60ms`) the reserved resources for this user will be released.
- `implicitReleaseAfter = 4`, after 4 empty SPS occasions (`4 * 20ms=80ms`) the reserved resources for this user will be released.
- `implicitReleaseAfter = 8`, after 8 empty SPS occasions (`8 * 20ms=160ms`) the reserved resources for this user will be released.

In order to understand the aforementioned parameters in depth, we shall examine a case where 6 Users start a voice call within the next 10ms, in an LTE network. When the call starts for each User, Semi Persistent Scheduling is activated with:

- `periodicity=20ms`, meaning that every 20ms there will be a reserved Resource Block for Users 1-6, so that they can transmit an audio packet in the uplink direction.
- `implicitReleaseAfter=2`, meaning that if two consecutive reserved Resource Blocks do not contain any audio data, these resources of this user are released.

As we can see in the Figure 40, the users of the network are distributed among the 4 Uplink subframes of the first frame that configuration 1 offers, according to the time they started the call.

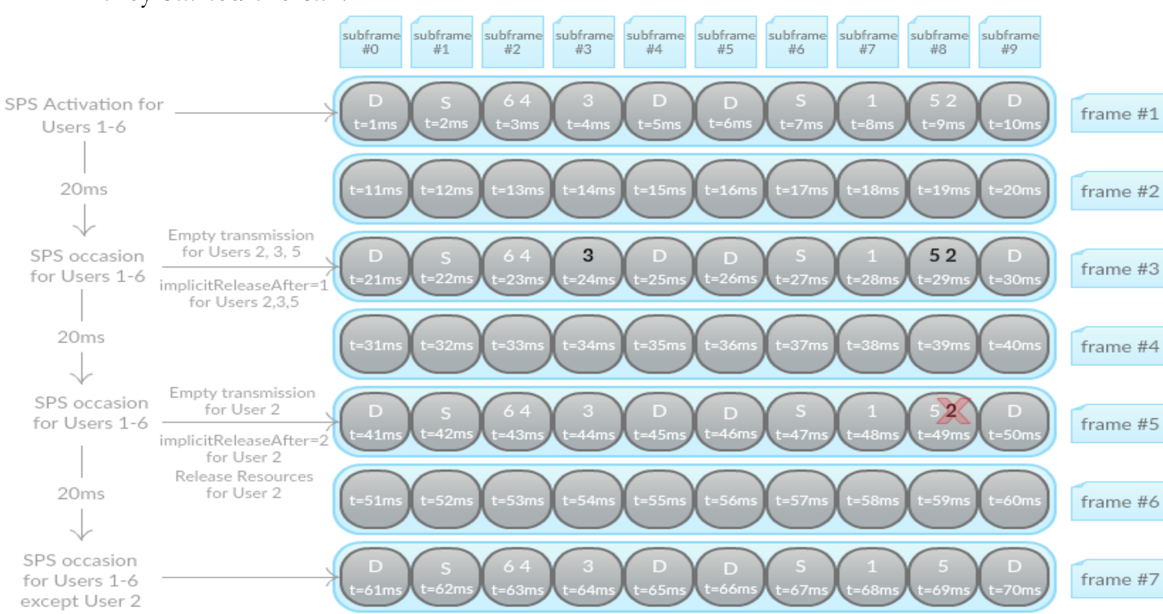



*Figure 40.* `implicitReleaseAfter` parameter operation example

By the time Semi-Persistent scheduling is activated for each one of the 6 Users in frame #1, every second frame to come will be an SPS occasion. Although, if a User does not transmit a payload `implicitReleaseAfter` consecutive times (`=2` in this example), such as User 2, the resources are released. As we can see, this User initiated Semi-Persistent scheduling algorithm at `t=9ms` and did not transmit any audio packets at `t=29ms` (`implicitReleaseAfter=1`) and `t=49ms` (`implicitReleaseAfter=2`). In other words, he didn't utilize two consecutive Semi-Persistent scheduling occasions, resulting to the release of his resources because the algorithm assumes that he is silent. Meanwhile, User 5 who started being scheduled by SPS at the exact same time with User 2, did also not transmit an audio packet at `t=29ms` (`implicitReleaseAfter=1`). However, at `t=49ms` User 5 managed to correct that, meaning that he gets to keep his reserved resources, cause `implicitReleaseAfter` counter resets its value every time we encounter a successful transmission.

To understand how significant the `implicitReleaseAfter` counter is for SPS, let's make clear that in real world scenarios the audio packets generation time interval is not fixed to 20ms, because link conditions change. For example, a fluctuation of 5ms in the packet generation process is very reasonable, whilst in extreme situations an audio packet might be ready for transmission after 40ms. Moreover, packet loss is a second factor that SPS has to deal with. Thus, if User 2 in the example presented above wasn't ready to transmit an audio packet until `t=69ms`, due to one of the reasons explained above, he would have to re-initiate Semi-Persistent Scheduling, which is not desirable in terms of control channel resource utilization. To overcome this problem, we just have to set `implicitReleaseAfter=3`, which gives the User one more opportunity to transmit an audio payload and consequently keep the reserved resources intact.

Although increasing the value of `implicitReleaseAfter` eliminates the chances to release the resources due to a false silence period prediction, simultaneously we increase the time by which our system predicts an actual silence period. For instance, if we set `implicitReleaseAfter=8` we definitely will not release a User's resources on account of a packet loss or bad link conditions, but we delay the prediction of the occurring silence period. On the other hand, if we set `implicitReleaseAfter=2`, packet loss and link conditions become the true obstacle to overcome, while the delay to predict an actual silence period is annihilated.

Conclusively, the value chosen for `implicitReleaseAfter` has a huge impact upon the operation of Semi-Persistent algorithm. Therefore, in the proposed scheme we utilize a decision tree learning algorithm that predicts the value with the ideal outcome for each of the users in the network that operate Semi-Persistent Scheduling [15]. The algorithm essentially considers each of the four values of the `implicitReleaseAfter` parameter as a distinctive class. Every time a user adds an audio packet to the queue, it computes the mean packet generation time. According to that value we are capable of choosing which class suits best for this particular user. For instance, if a user generates an audio packet every 25-30ms, `implicitReleaseAfter=2` is considered as the optimal value, whilst `implicitReleaseAfter=3` emerges as the mean values tend to 40ms.

The above prediction scheme enhances SPS because it conforms to each user's link conditions and talk activity pattern. Thereby, the algorithm maintains an advantageous position regarding the possibility to predict an actual silence period, while simultaneously the chances to falsely predict one are reduced. However, the capabilities of the decision tree learning algorithm that was chosen are not narrowed down to the aforesaid.

Decision tree learning, as a machine learning algorithm, is a statistical method not only used to produce reliable, repeatable decisions and results, but also uncover hidden insights through learning from historical relationships and trends in the data. Consequently, the iterative aspect is important because as models are exposed to new data, they are able to independently adapt.

The aforementioned prediction scheme is simplistically illustrated in Figure 40.

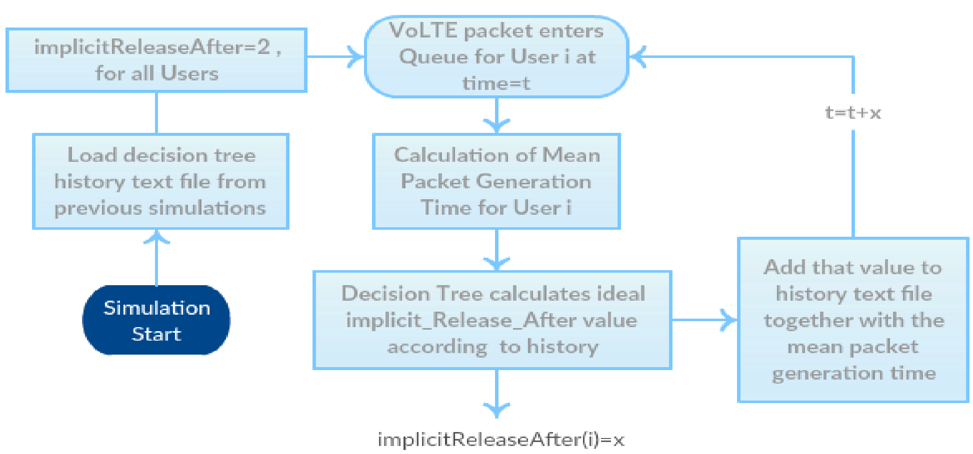



***Figure 40.*** Prediction schemes operation in a simulation

# 6. Modeling and Simulation Environment

## 6.1 Environment

For the evaluation of the abovementioned algorithms Matlab was utilized. The MATLAB platform is optimized for solving engineering and scientific problems. The matrix-based MATLAB language is the world's most natural way to express computational mathematics, which applies perfectly in the case of LTE. Built-in graphics make it easy to visualize and gain insights from data. A vast library of prebuilt toolboxes lets us get started right away with algorithms essential to the user's domain.

## 6.2 LTE Frame Construction

For the LTE Physical Layer implementation, the LTE System Toolbox was utilized. LTE System Toolbox provides standard-compliant functions and apps for the design, simulation, and verification of LTE communications systems. The system toolbox accelerates LTE algorithm and physical layer development.

The standard-compliant functions provided by this toolbox are built in agreement with ETSI. ETSI produces a range of specifications, standards, reports and guides, each with its own particular purpose. Collectively they are all called standards. The ability to produce these different types of documents meets the variety of needs, including Long Term Evolution.

For our purpose, a specific type of ETSI standard was used, called Technical Specification (TS). ETSI TS documents contain technical requirements approved by the Technical Committee that drafted each technology. In Figure 41, we can see the code that produces the LTE frame according to TS 36 series, which exposes the LTE technical information.

```
function [PDCCH,DCI,rmc,frc] = TDDFrame()
frc=lteRMCUL('A4-6','TDD');          36.104 document
rmc=lteRMCDL('R.3','TDD');           36.101 document
PDCCH=ltePDCCHInfo(rmc);             36.211 document
DCI=lteDCIInfo(rmc);                 36.212 document
end
```

***Figure 41***. LTE frame using LTE System Toolbox

## 6.2.1 Downlink Subframes

The lteRMCDL() function returns a configuration structure for the downlink reference channel defined as 'R.3', using the 'TDD' duplexing mode. This structure uses a channel-specific default configuration and contains all the configuration parameters required for our simulation of the forward link. These parameters are shown in Figure 42.

| Field | Value |
|---|---|
| RC | 'R.3' |
| NDLRB | 50 |
| CellRefP | 1 |
| NCellID | 0 |
| CyclicPrefix | 'Normal' |
| CFI | 2 |
| Ng | 'Sixth' |
| PHICHDuration | 'Normal' |
| NFrame | 0 |
| NSubframe | 0 |
| TotSubframes | 10 |
| OCNG | 'Off' |
| Windowing | 0 |
| DuplexMode | 'TDD' |
| PDSCH | *1x1 struct* |
| SSC | 4 |
| TDDConfig | 1 |

***Figure 42.*** Configuration Parameters of downlink using LTE System Toolbox

The field names and default values comply with the definition found in TS 36.101, Annex A.3. Thereby, each field in the figure above is produced in reflection to the 'R3' reference channel, as illustrated in Table 14.

Moreover, all the informations regarding the Physical Downlink Control Channel were obtained through the ltePDCCHInfo() function, using the configuration parameters provided by the lteRMCDL() as an input. The output of this function is shown in Figure 43.

| Field | Value |
|---|---|
| NREG | 246 |
| NRE | 984 |
| NCCE | 27 |
| NREGUsed | 243 |
| NREUsed | 972 |
| MTot | 1968 |
| NSymbols | 2 |

***Figure 43.*** Configuration parameters of PDCCH using LTE System Toolbox

The first NSymbols OFDM symbols represent the control region of each subframe and carry the PCFICH, PHICH and PDCCH, while the MTot field contains the capacity of PDCCH in bits. These informations are valuable because they provide the resources that will be used to transfer the encoded DCI message.

Last but not least, the size in bits of each type of DCI message is brought to us through the lteDCIInfo() function, using the configuration parameters provided by lteRMCDL() as an input, once again. The output of this function is shown in Figure 44.

| Field | Value |
| --- | --- |
| Format0 | 29 |
| Format1 | 34 |
| Format1A | 29 |
| Format1B | 31 |
| Format1C | 13 |
| Format1D | 31 |
| Format2 | 46 |
| Format2A | 43 |
| Format2B | 43 |
| Format2C | 46 |
| Format2D | 48 |
| Format3 | 29 |
| Format3A | 29 |
| Format4 | 41 |

*Figure 44.* DCI bits per Format

A plain representation of the informations we gathered from each function used, is illustrated in Figure 45.

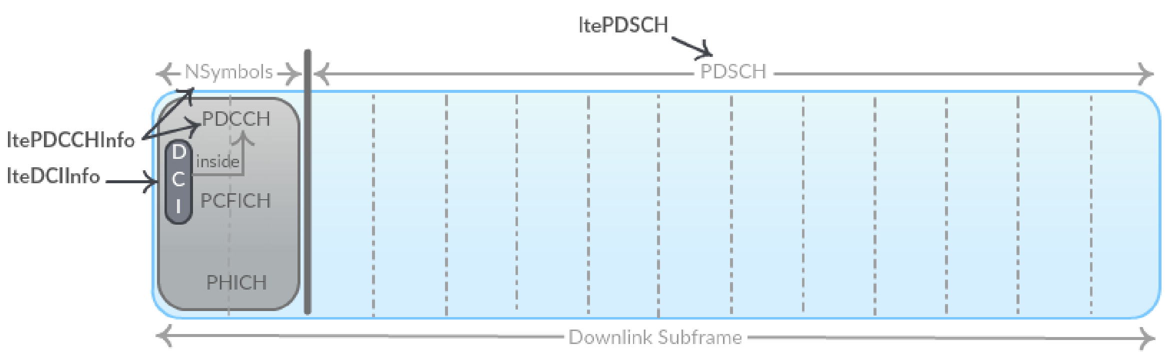



*Figure 45.* Gathered Informations by LTE System Toolbox

### 6.2.2 Uplink Subframe

The lteRMCUL function returns a configuration structure for the reference channel defined as 'R.4-6', using the 'TDD' duplexing mode. This structure contains all the configuration parameters required for our simulation of the reverse link. These parameters are shown in Figure 46.

| Field | Value |
|---|---|
| RC | 'A4-6' |
| NULRB | 50 |
| NCellID | 0 |
| NFrame | 0 |
| NSubframe | 0 |
| CyclicPrefixUL | 'Normal' |
| CyclicShift | 0 |
| Shortened | 0 |
| Hopping | 'Off' |
| SeqGroup | 0 |
| TotSubframes | 10 |
| RNTI | 1 |
| NTxAnts | 1 |
| Windowing | 0 |
| DuplexMode | 'TDD' |
| PUSCH | *1x1 struct* |
| SSC | 0 |
| TDDConfig | 1 |

***Figure 46.*** Configuration Parameters of uplink using LTE System Toolbox

The field names and default values comply with the definition found in TS 36.104, Annex A. Thereby each field in the figure above is produced in reflection to the 'A4-6' fixed reference channel, as illustrated in Table 19.

***Table 19.*** Reference Measurement Channel configuration parameters for uplink

| Parameter | Unit | Value |
|---|---|---|
| Reference Channel | | A4-6 |
| Allocated Resource Blocks | | 50 |
| Modulation | | 16QAM |
| Code Rate | | 3/4 |
| Payload Size | Bits | 21384 |

## 6.3 Traffic

The uplink traffic of the network was implemented using real traffic traces captured while performing the following services:

- VoLTE call
- Live Streaming
- Real Video

The information that we gathered by capturing these types of traffic were the packet bits and the time interval within which the packet entered the queue, during the whole operation. Subsequently, using a probability density function we described the relative likelihood for each variable to take on a given value. The results of this function for each type of traffic are shown below.

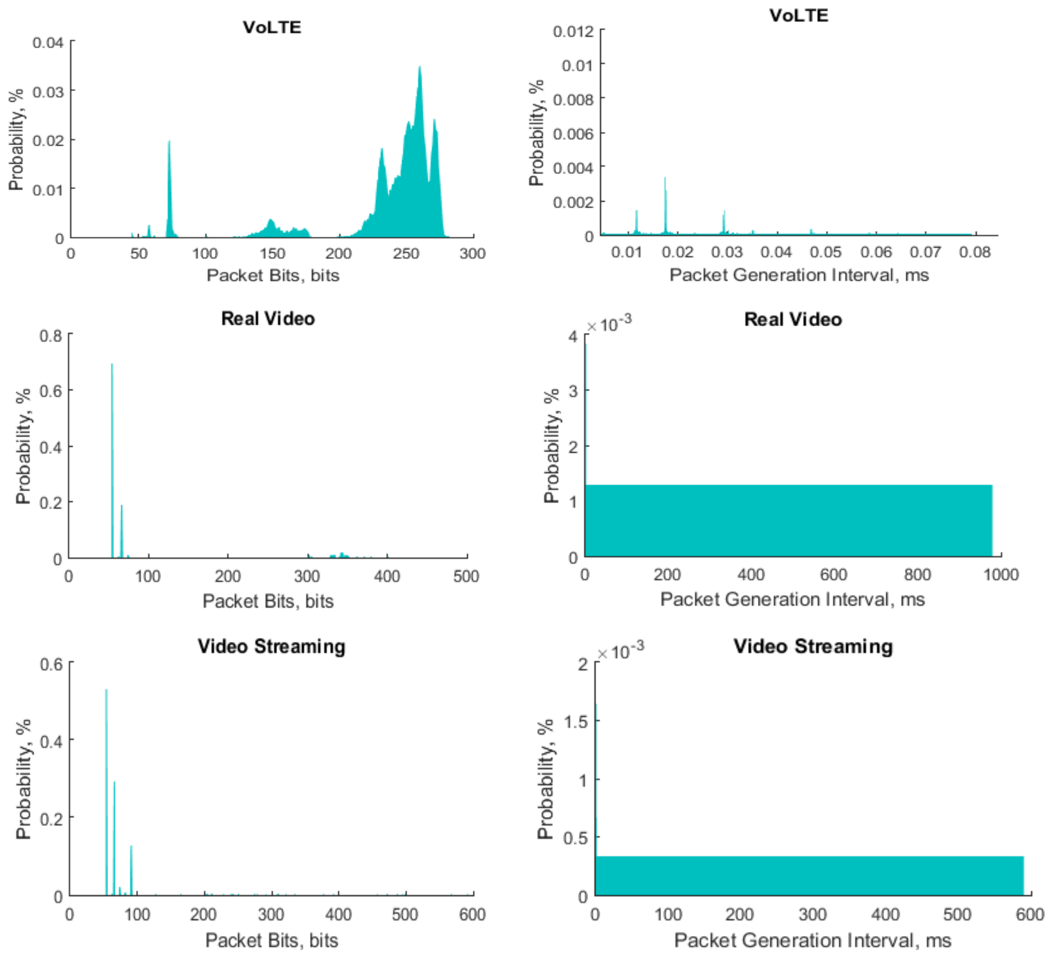



***Figure 47.*** Traffic bits and Mean Packet Generation Intervals

In the downlink direction we use exponential distribution to simulate the traffic produced by VoLTE, Real Video and Video Streaming. Each of these services is assigned to a priority level, meaning that in case the network encounters congestion, the lowest priority level traffic would be the first to be discarded.

# 6.4 Scheduling Algorithms

## 6.4.1 Background

The appliance of the LTE System Toolbox functions essentially gave us all the required information about the capacity of each channel inside each and every LTE subframe. Thus, we know the amount of resources that can be scheduled every millisecond to the users of our simulation network. Moreover, the probability density function provides the packet bits and the interval within which every user inside our network adds a new packet to the queue. All that is left is to add the scheduling algorithms that handles the traffic.

In the context of this thesis, two scheduling algorithms were implemented. These algorithms are the following:

- Dynamic Scheduling, suitable for bursty, unpredictable and download type of traffic, such as Real Video and Video Streaming.
- Semi-Persistent Scheduling, suitable for traffic defined by short bursts, while at regular intervals, such as VoLTE.

Additionally, FIFO was also carried out to compare its performance with the other two scheduling schemes.

## 6.4.2 Dynamic Scheduling

This scheduling method utilizes the following input parameters:

- $\lambda_{BS}$ , which directly affects the packet generation ratio in the forward link.
- `Sim_Time`, which sets the duration of the simulation.
- $P_1$, which defines the amount of subframe resources that will be used by `priority=1` traffic, meaning VoLTE.

- `P₂`, which defines the amount of subframe resources that will be used by `priority=2` traffic, meaning Real Video.
- `P₃`, which defines the amount of subframe resources that will be used by `priority=3` traffic, meaning Video Streaming.

Consequently, Figure 48 illustrates the way Dynamic Scheduling grants the uplink packets in every downlink subframe occurrences, starting from the first subframe of each frame. While the allocation of the uplink packets takes place in the downlink subframes, uplink subframes empty the queue of the users accordingly.

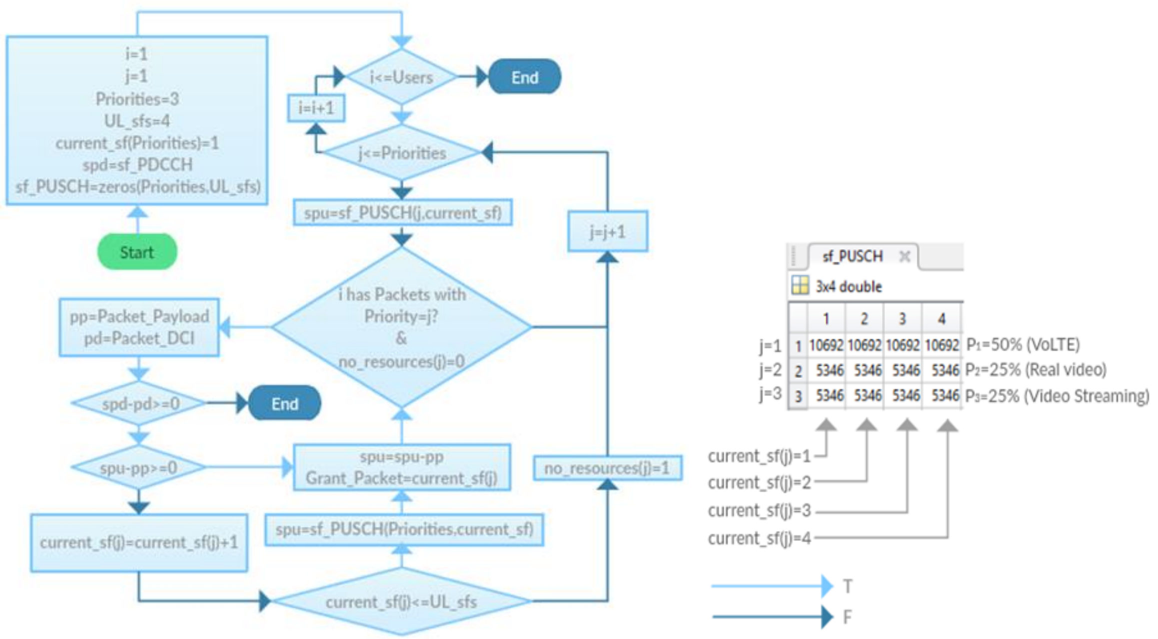



*Figure 48.* Flowchart of DS algorithm

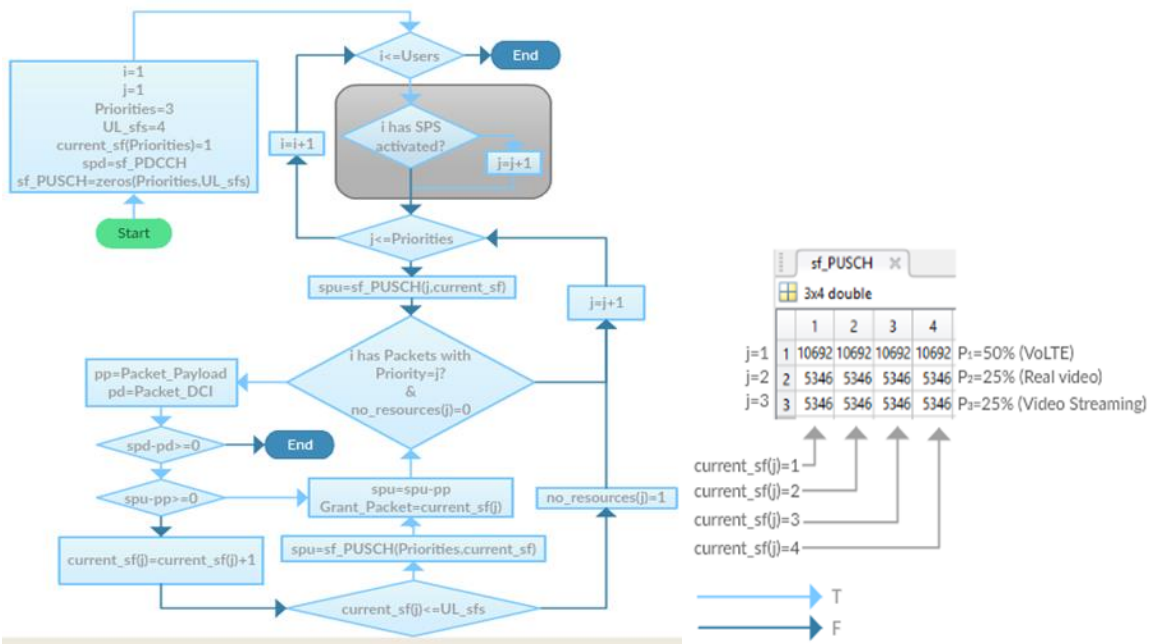
### 6.4.3 Semi-Persistent Scheduling

This scheduling method utilizes the following input parameters:

- $\lambda_{BS}$ , which directly affects the packet generation ratio in the forward link.
- `Sim_Time`, which sets the duration of the simulation in seconds.
- `P₁`, which defines the amount of subframe resources in percentage, that will be used by `priority=1` traffic, meaning VoLTE.
- `P₂`, which defines the amount of subframe resources in percentage, that will be used by `priority=2` traffic, meaning Real Video.

- `P3`, which defines the amount of subframe resources in percentage, that will be used by `priority=3` traffic, meaning Video Streaming.
- `implicitReleaseAfter`, which defines the number of empty transmissions after which SPS gets deactivated for the current User.

The following flowchart illustrates the way that Semi Persistent Scheduling grants the uplink packets in every downlink subframe occurrences, starting from the first subframe of each frame.




| | 1 | 2 | 3 | 4 |
|---|---|---|---|---|
| 1 | 10692 | 10692 | 10692 | 10692 |
| 2 | 5346 | 5346 | 5346 | 5346 |
| 3 | 5346 | 5346 | 5346 | 5346 |

***Figure 49.*** Flowchart of SPS algorithm

The differences between the flowchart of DS and SPS are grayed out. It is important to note that SPS handles only the VoLTE traffic, while DS manages Real Video and Video Streaming. Accordingly, if a user has already activated SPS, all the VoLTE traffic is ignored by DS. In the case where SPS is not yet activated for the current user, DS allocates the first audio packet and as of now SPS is considered activated.

### 6.4.4 Proposed Semi-Persistent Scheduling

This proposed scheduling method utilizes the same input parameters described in the above section, excluding `implicitReleaseAfter`. The reason behind this lies to the fact that we are capable of computing the ideal value for this variable thanks to the prediction algorithm we implemented, as described in the previous chapter. Concluding, the flowchart remains the same, because the prediction algorithm schedules the packets in the exact same way.

# 7. Evaluation & Numerical Results

In this chapter we examine the efficiency of the following algorithms in several case scenarios:

- FIFO Scheduling
- Dynamic Scheduling
- Semi-Persistent Scheduling
- Proposed Semi-Persistent Scheduling

The efficiency of the above scheduling techniques is shown in terms of mean packet delay, jitter and throughput. Finally, the number of users in the simulations network was set to 100, 200, 300, 400 and 500, whilst the duration of the simulation was set to 10 seconds.

## 7.1 First case scenario

In the first case scenario we consider that in each downlink and uplink subframe:

- 50% of the resources are available for VoLTE scheduling.
- 25% of the resources are available for Real Video scheduling.
- 25% of the resources are available for Video Streaming scheduling.

However, in the case of Semi-Persistent Scheduling and the proposed edition of that technique, VoLTE traffic is prioritized over the resource allocation scheme discussed above. Thereby, the users that have already initiated a VoLTE call are predominantly allocated so that their QoS need is satisfied.

### 7.1.1 Mean Packet Delay

The following figures illustrate the mean packet delay of each scheduling method in relation to the number of users in the network. The title of each figure specifies the scheduling technique that was applied. Moreover, in the case of Semi-Persistent Scheduling a variable named "Im" is defined, referring to the value of the `implicitReleaseAfter` parameter that was discussed in the chapters above. The last figure represents the proposed SPS algorithm that conforms the value of `implicitReleaseAfter` according to each user's needs.

What becomes comprehensible from Figure 50, is that FIFO is inefficient for large number of users. This kind of behavior was expected, because this scheduling technique does not take into account the priorities of each traffic type. Thus, when congestion occurs, the network has no mechanics to deal with it, leading to excessive delay values.

Subsequently, Dynamic Scheduling seems appropriate for Real Video and Video Streaming traffic that are bursty and unpredictable, as seen in Figure 50. On the other hand, DS inefficiency becomes evident in the case of VoLTE traffic, because packets are relatively small and arrive at regular intervals.

It is also noteworthy that DS behaves in a similar way to FIFO for massive amounts of VoLTE users in the network. This correlation between these seemingly distinctive scheduling schemes is reasonal, because the control channel is setting a threshold as to how many users can be allocated. Therefore, even if we had 100% of the subframe resources assigned to the VoLTE users, they would still be unsatisfied.

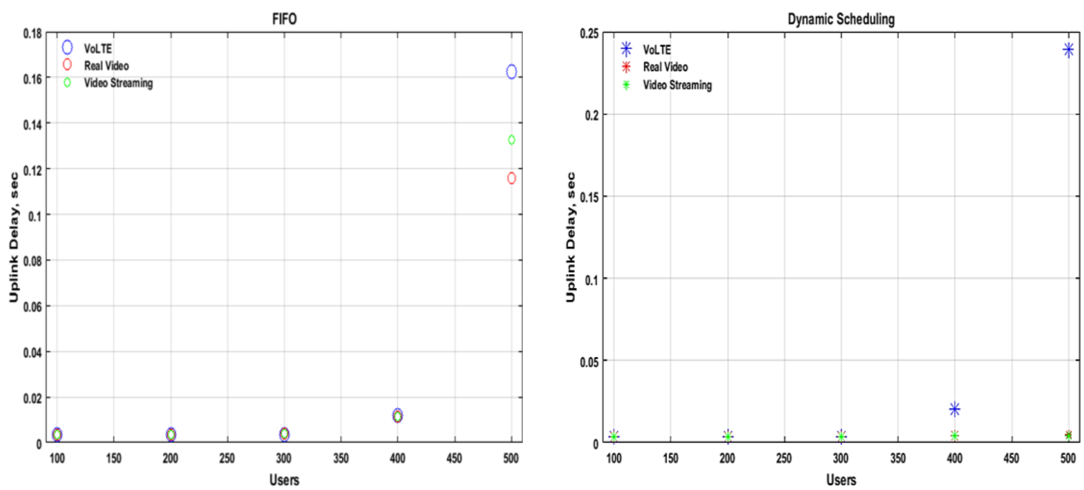



***Figure 50.*** FIFO and Dynamic Scheduling Mean Packet Delays per Traffic

Semi-Persistent scheduling addresses the aforementioned inability of DS, by exploiting the fairly consistent and predictable transmission pattern of VoLTE packets. Thus the combination of DS and SPS produces the results shown in Figure 51.

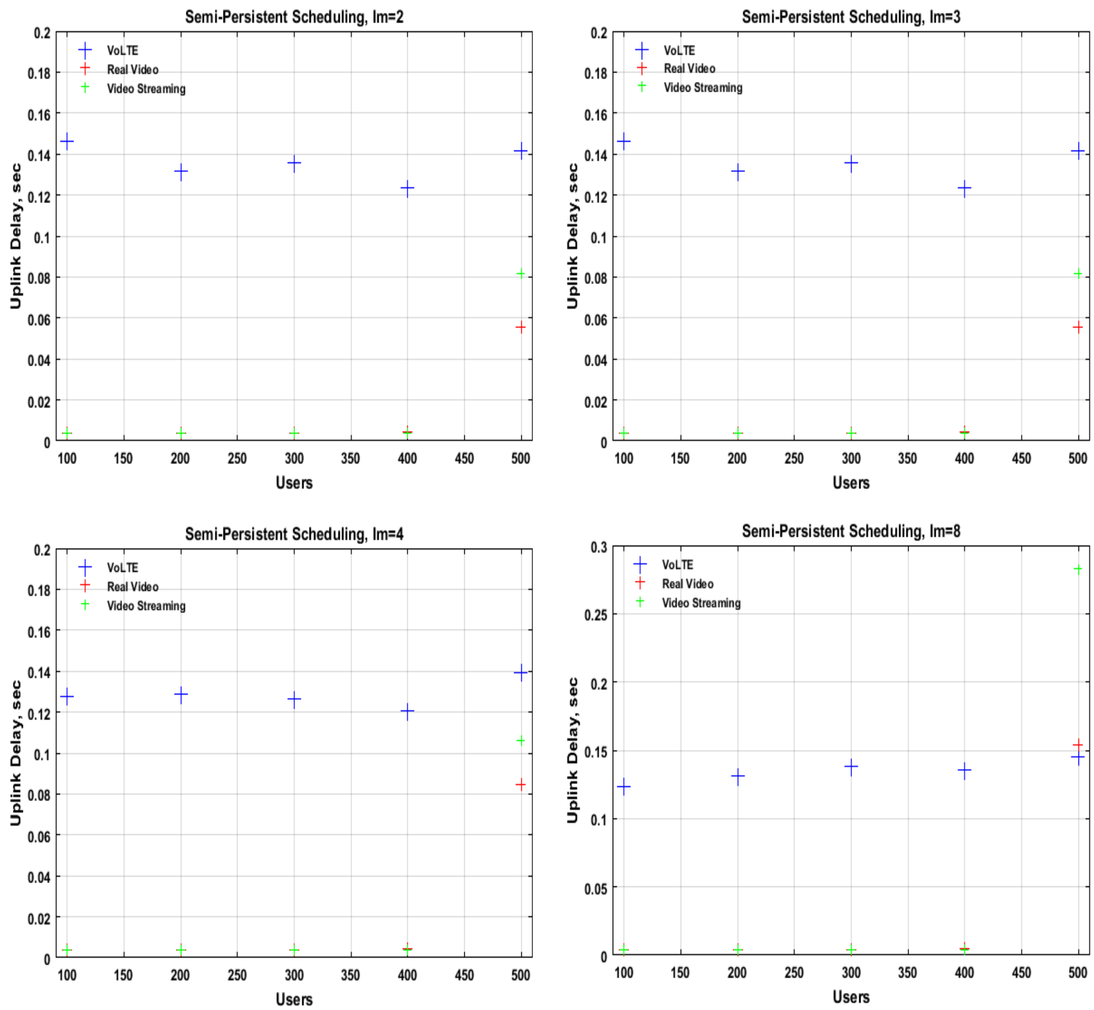



*Figure 51.* Semi-Persistent Scheduling Mean Packet Delays per Traffic

At first glance, one might say that DS or FIFO seems better at handling VoLTE traffic for low number of users, because delay is lower. However that statement is not considered true, because SPS maintains a constant acceptable VoLTE delay value regardless of the number of users, while simultaneously it decongests the control channel allowing variant traffic types to be scheduled dynamically. That becomes noticeable at the highest number of users. In that instance, DS is incapable of keeping a sufficient VoLTE delay, due to the lack of PDCCH resources.

Moreover, it is evident that the value of the `implicitReleaseAfter` variable affects the mean packet dealy value of Real Video and Video Streaming in Figure 51. That is a fact because high `implicitReleaseAfter` values mean that the algorithm delays proportionally the prediction of an upcoming silence period. This delay influences the release of the resources for each VoLTE user. Thereby, when congestion occurs in the network, the other traffic types fail to be scheduled on time due to that resource waste, leading to high mean delay values.

That being said, it is proven that the coexistence of DS and SPS provides the optimal solution to the significant problem of latency reduction. However, adjusting the `implicitReleaseAfter` parameter according to each users mean packet generation interval might provide even better results. Following the above notion, we built an enhanced version of the SPS mechanism that provides even better results as shown below.

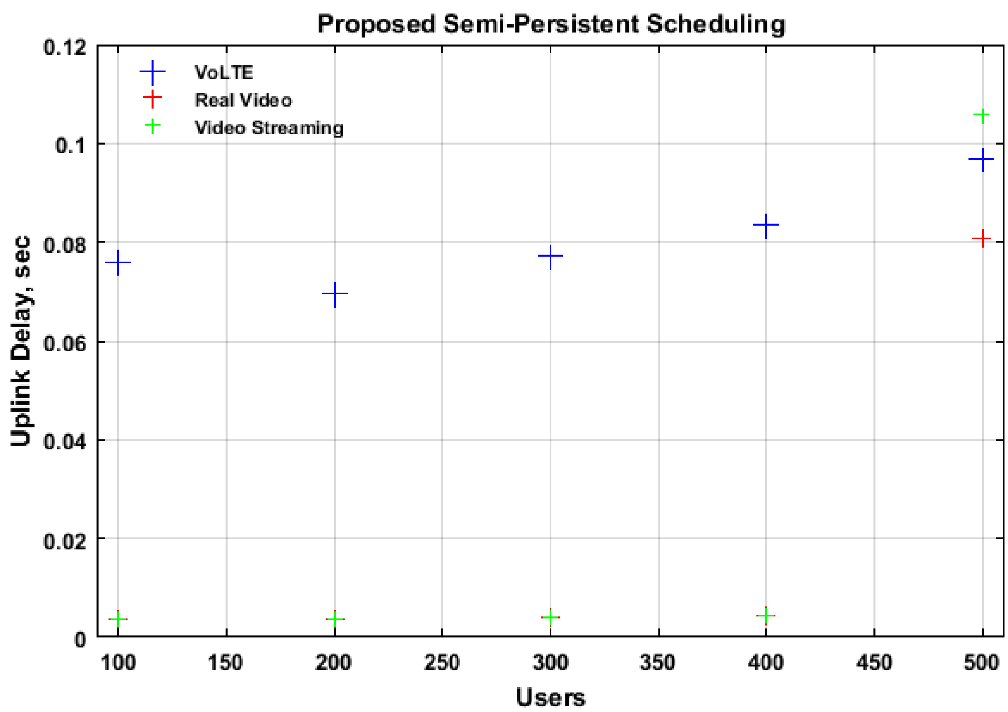



*Figure 52.* Proposed Semi Persistent Scheduling Algorithm Mean Packet Delays per Traffic

## 7.1.2 Jitter

The following figures illustrate the measured jitter of each scheduling method in relation to the number of users in the network. The title of each figure specifies the scheduling technique that was applied. Moreover, in the case of Semi-Persistent Scheduling a variable named "Im" is defined, referring to the value of the `implicitReleaseAfter` parameter that was discussed in the chapters above. The last figure represents the proposed SPS algorithm that conforms the value of `implicitReleaseAfter` according to each user's needs.

The behavior of jitter is directly influenced by mean packet delay. Therefore, the above statements about the efficiency of each scheduling method regarding delay are applicable to this case to.

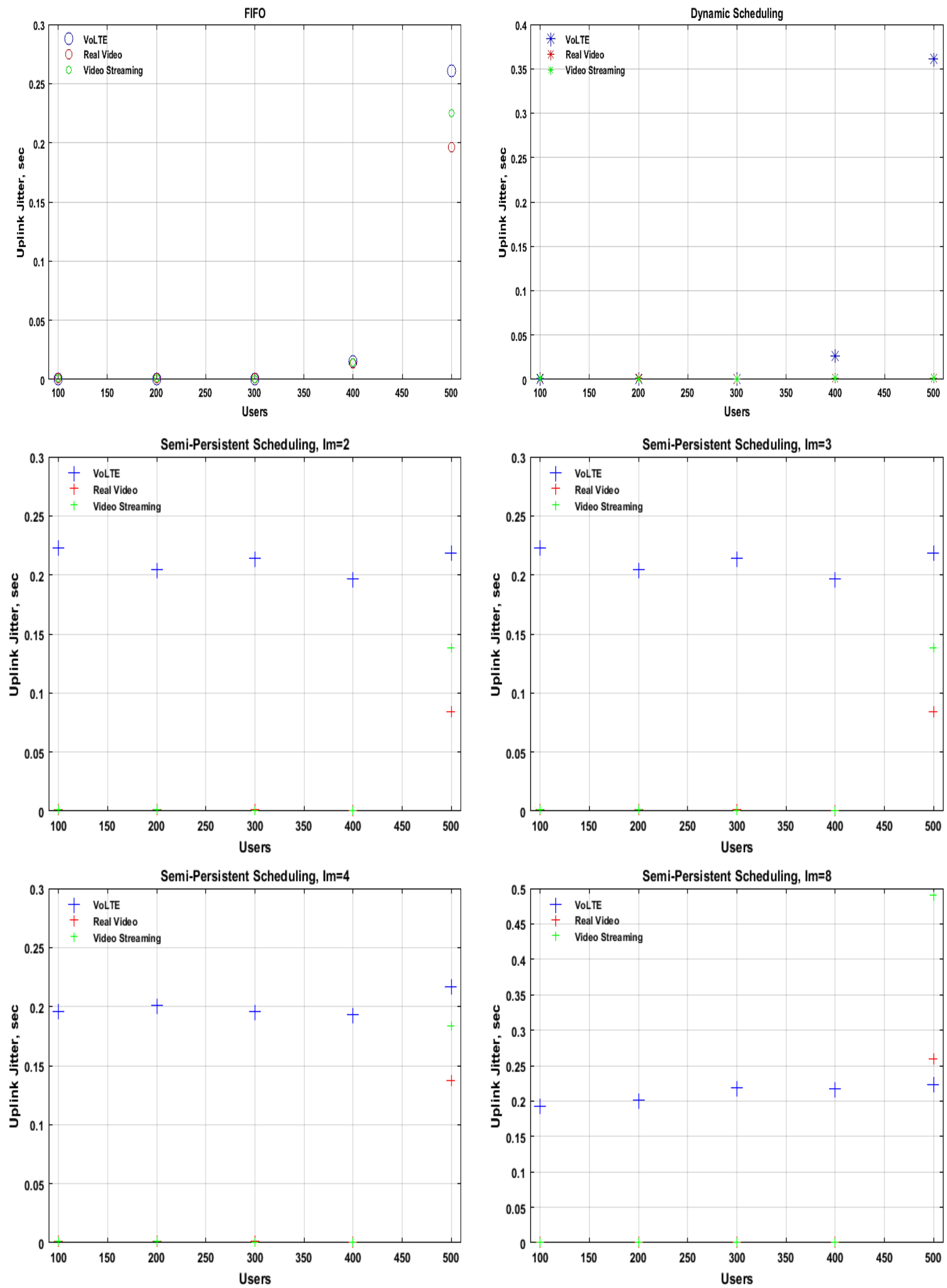



*Figure 53.* FIFO, Dynamic and Semi Persistent Scheduling Algorithm Jitter per Traffic

In comparison to the algorithms of Figure 53, the Proposed Semi Persistent Scheduling algorithm provides better results, as depicted in the following figure.

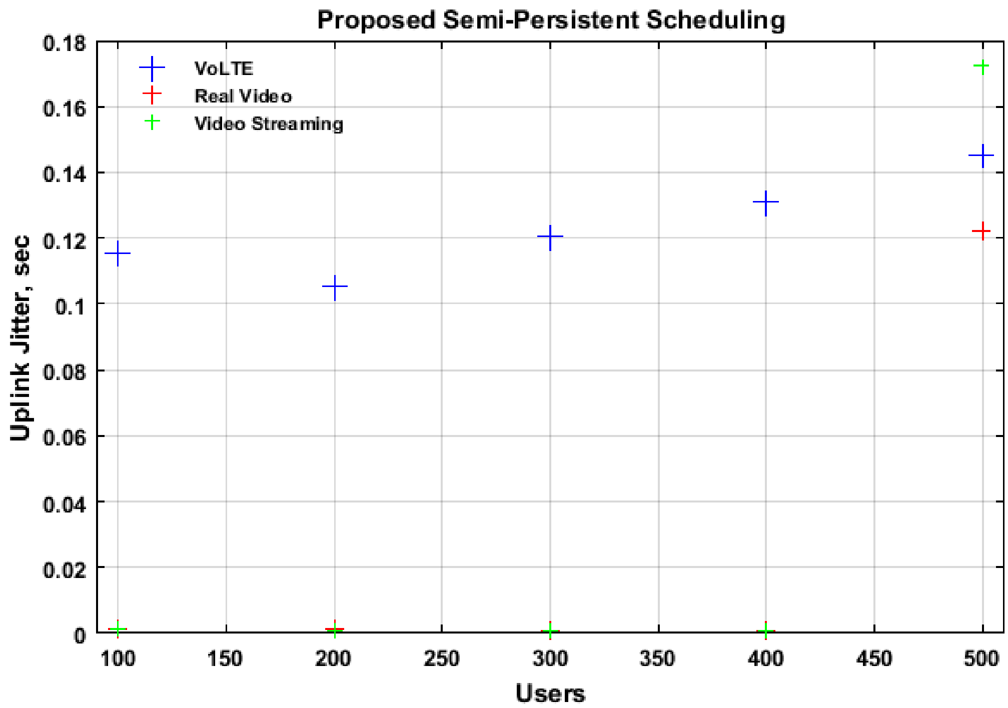



***Figure 54.*** Proposed Semi Persistent Scheduling Algorithm Jitter per Traffic

### 7.1.3 Throughput

Figure 55 illustrates the throughput of each scheduling method in relation to the number of users in the network. The title of each figure specifies the scheduling technique that was applied. Moreover, in the case of Semi-Persistent Scheduling a variable named "Im" is defined, referring to the value of the `implicitReleaseAfter` parameter that was discussed in the chapters above. The last figure represents the proposed SPS algorithm that conforms the value of `implicitReleaseAfter` according to each user's needs.

While throughput might be a decisive parameter in the examination of network efficiency, this is not the case. The reason behind this is that the traffic types used in this simulation are not that bandwidth consuming as control limited, especially VoLTE. Therefore, our main concern is the number of simultaneous voice calls that can be handled by PDCCH, while secure delay levels are maintained overall.

The above speculation contributes to the similarity observed in Figure 55, among all the scheduling schemes that were examined.

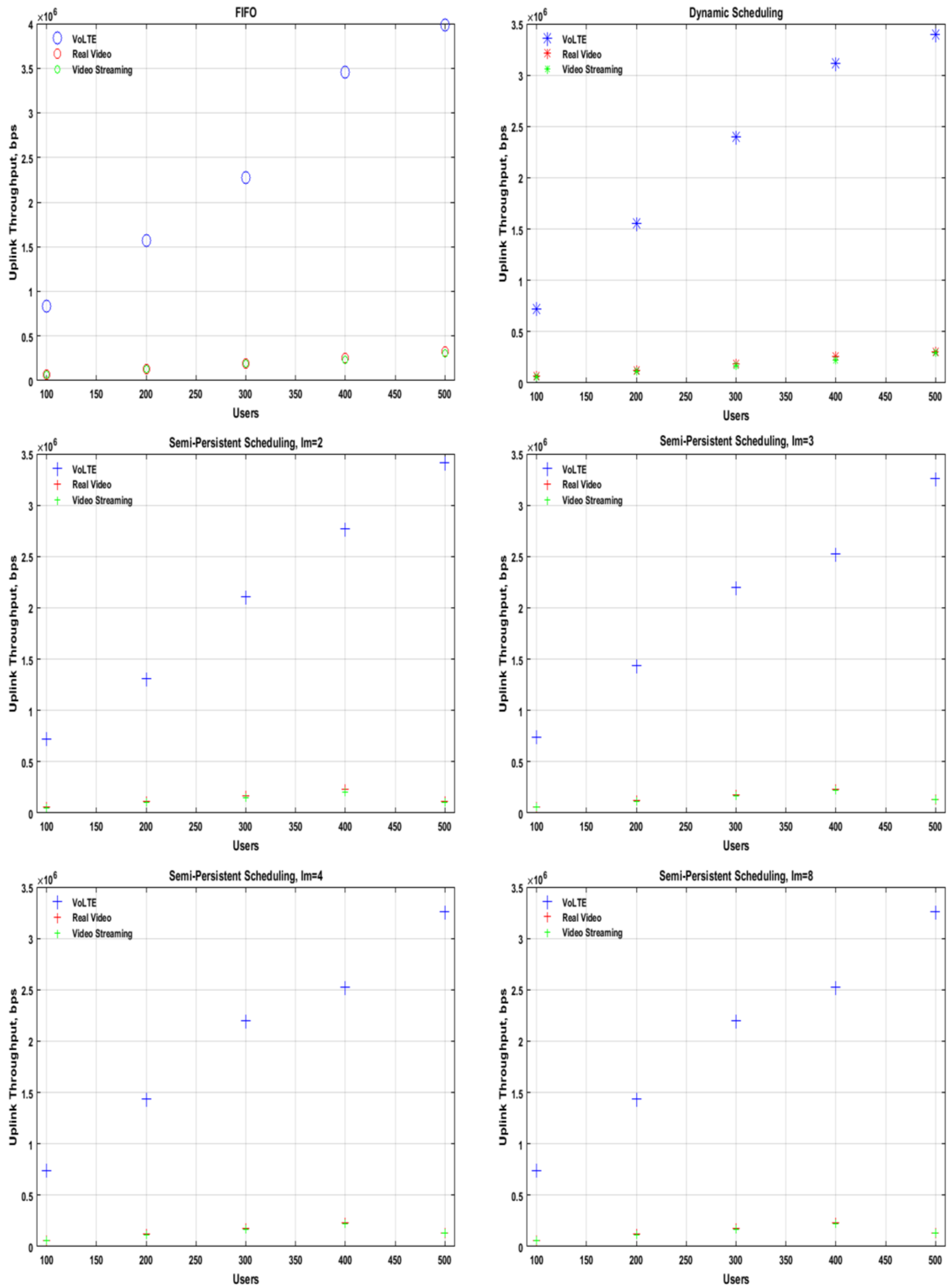



***Figure 55.*** FIFO, Dynamic and Semi Persistent Scheduling Algorithm Throughput per Traffic

It is highly important to mention that as the number of user's increments in FIFO and DS, throughput tends to stay at the same levels until it reaches a threshold. That threshold is set by the limited capacity of the control region, which in turn limits the number of packets that can be allocated in each subframe.

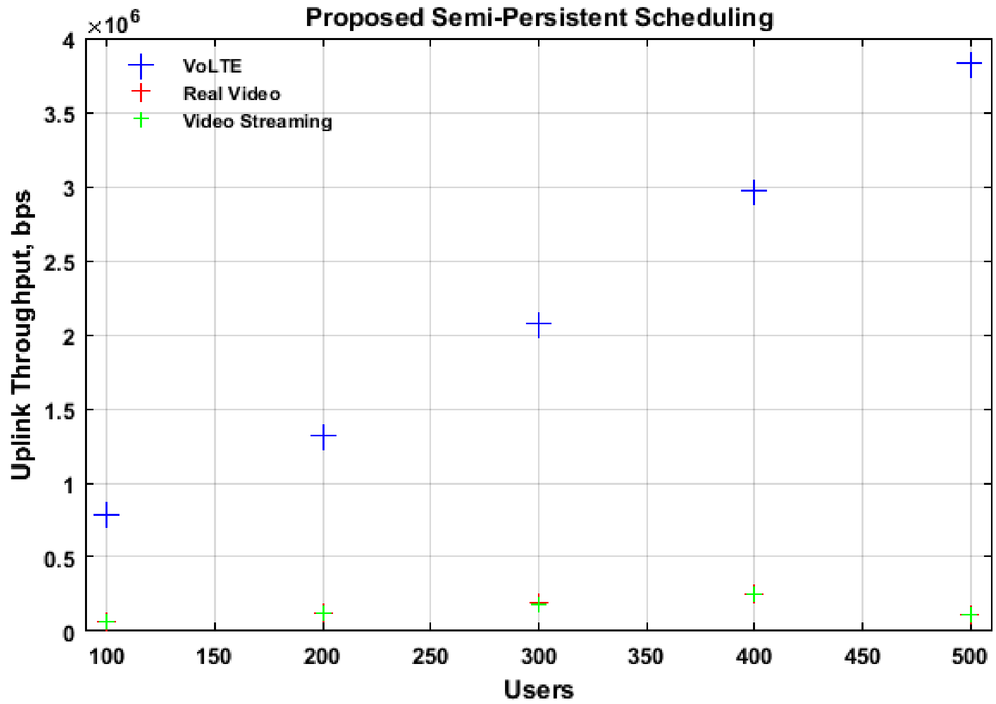



*Figure 56.* Proposed Semi Persistent Scheduling Algorithm Throughput per Traffic

## 7.2 Second Case Scenario (20 40 40)

In the second case scenario we consider that in each downlink and uplink subframe:

- 20% of the resources are available for VoLTE scheduling.
- 40% of the resources are available for Real Video scheduling.
- 40% of the resources are available for Video Streaming scheduling.

However, in the case of Semi-Persistent Scheduling and the proposed edition of that technique, VoLTE traffic is prioritized over the resource allocation scheme discussed above. Thereby, the users that have already initiated a VoLTE call are predominantly allocated so that their QoS need is satisfied.

This case scenario differs from the first one because the resources assigned to the VoLTE users are considered low for high numbers of users. Moreover, the increased control channel overhead still persists as a problem, because 20% of voice packet load in every subframe is still able to cause PDCCH overflow.

### 7.2.1 Delay

First things first, the below figure proves that FIFO is catastrophic in terms of latency, which is provoked by this techniques time-based packet scheduling scheme. Moreover the low resource availability for VoLTE plus the increased PDCCH overhead caused by this traffic, does not only affect VoLTE performance, but the other traffics too. That is the reason why Real Video and Video Streaming present high latency values, even when the resources allocated to them are sufficient.

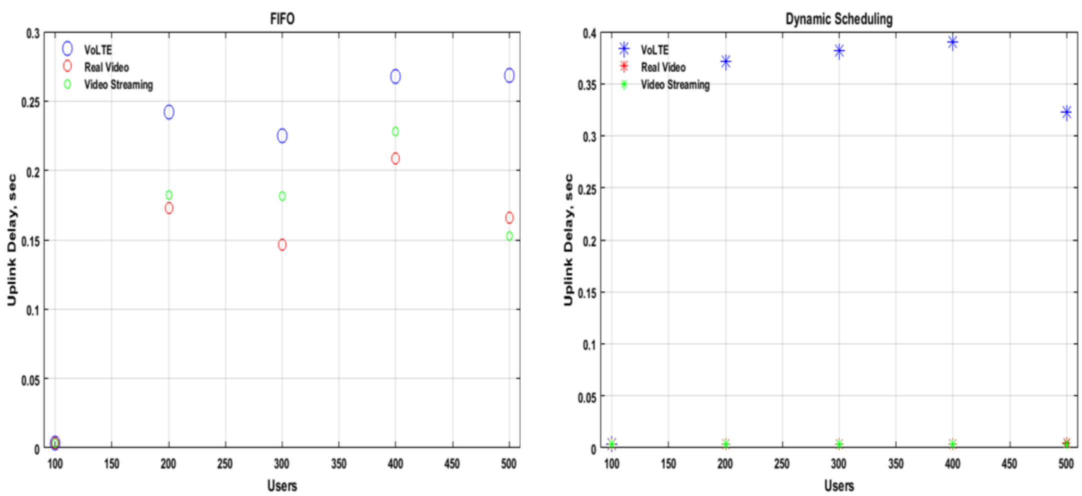



*Figure 57.* FIFO and Dynamic Scheduling Mean Packet Delay per Traffic

Following, DS performance regarding Real Video and Video Streaming meliorates in relation to FIFO, because this technique forwards the packets according to their priority. However that comes in the cost of even more VoLTE latency, due to the confined resources provided to this type of traffic and the control region overhead.

In the case of Semi-Persistent Scheduling, the control region overhead issue is settled, resulting to the enhanced VoLTE latency values shown below. The steep growth in the case of 500 users is considered normal due to the absence of resources.

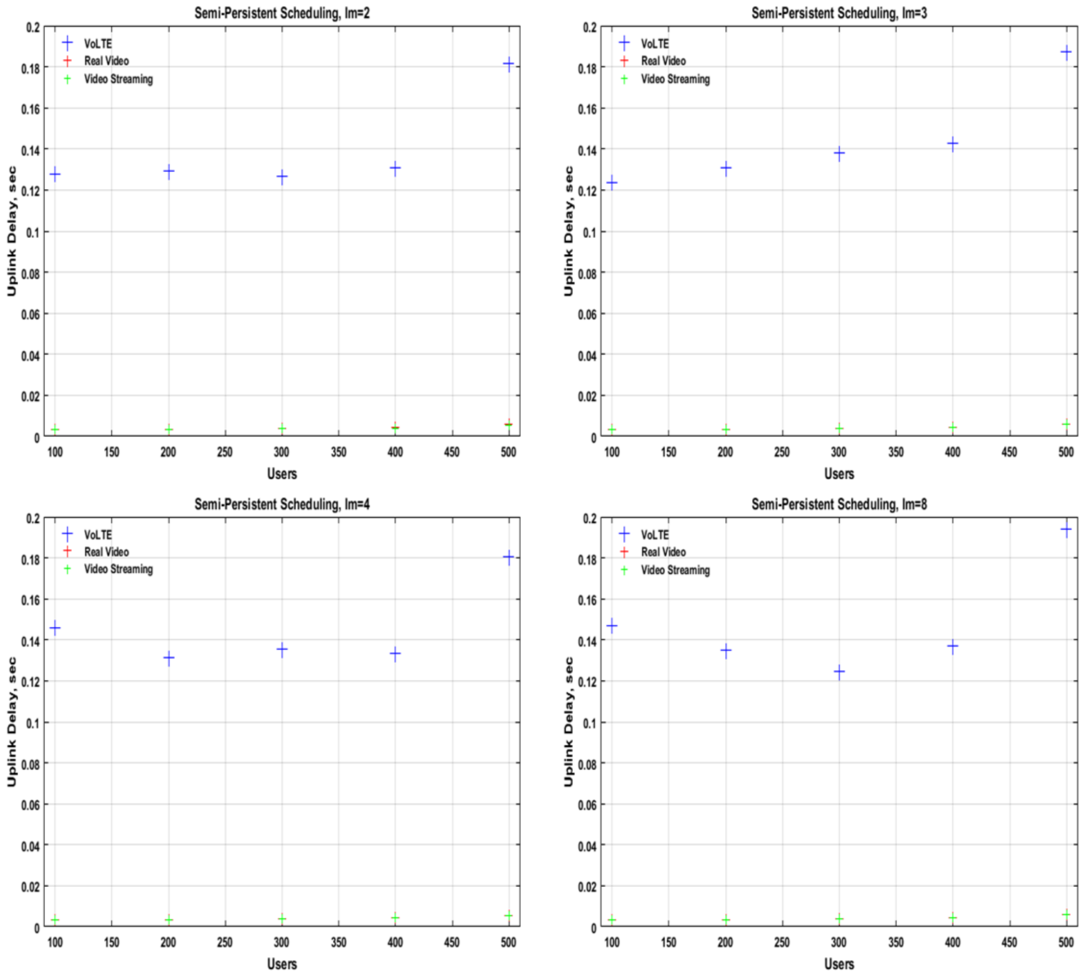



*Figure 58.* Semi Persistent Scheduling Mean Packet Delay per Traffic

Conclusively, the proposed scheme outperforms all the other scheduling techniques in terms of delay.

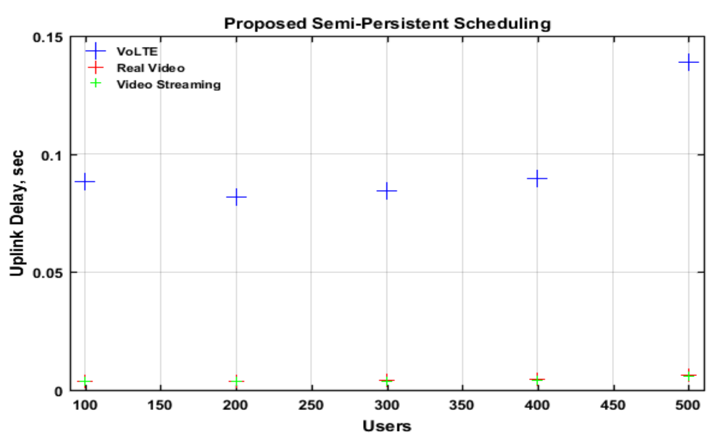



*Figure 59.* Proposed Semi Persistent Scheduling Mean Packet Delay per Traffic

### 7.2.2 Jitter

The following figures illustrate the jitter of each scheduling method in relation to the number of users in the network. The title of each figure specifies the scheduling technique that was applied. Moreover, in the case of Semi-Persistent Scheduling a variable named "Im" is defined, referring to the value of the `implicitReleaseAfter` parameter that was discussed in the chapters above. The last figure represents the proposed SPS algorithm that conforms the value of `implicitReleaseAfter` according to each user's needs.

The behavior of jitter is directly influenced by mean packet delay. Therefore, the above statements about the efficiency of each scheduling method regarding Delay are applicable to this case to.

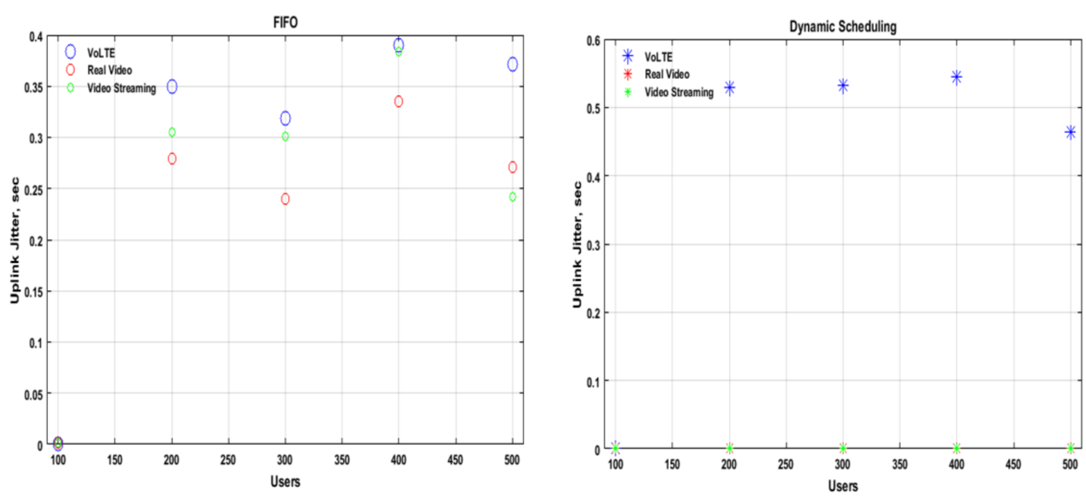



*Figure 60.* FIFO and Dynamic Scheduling Jitter per Traffic

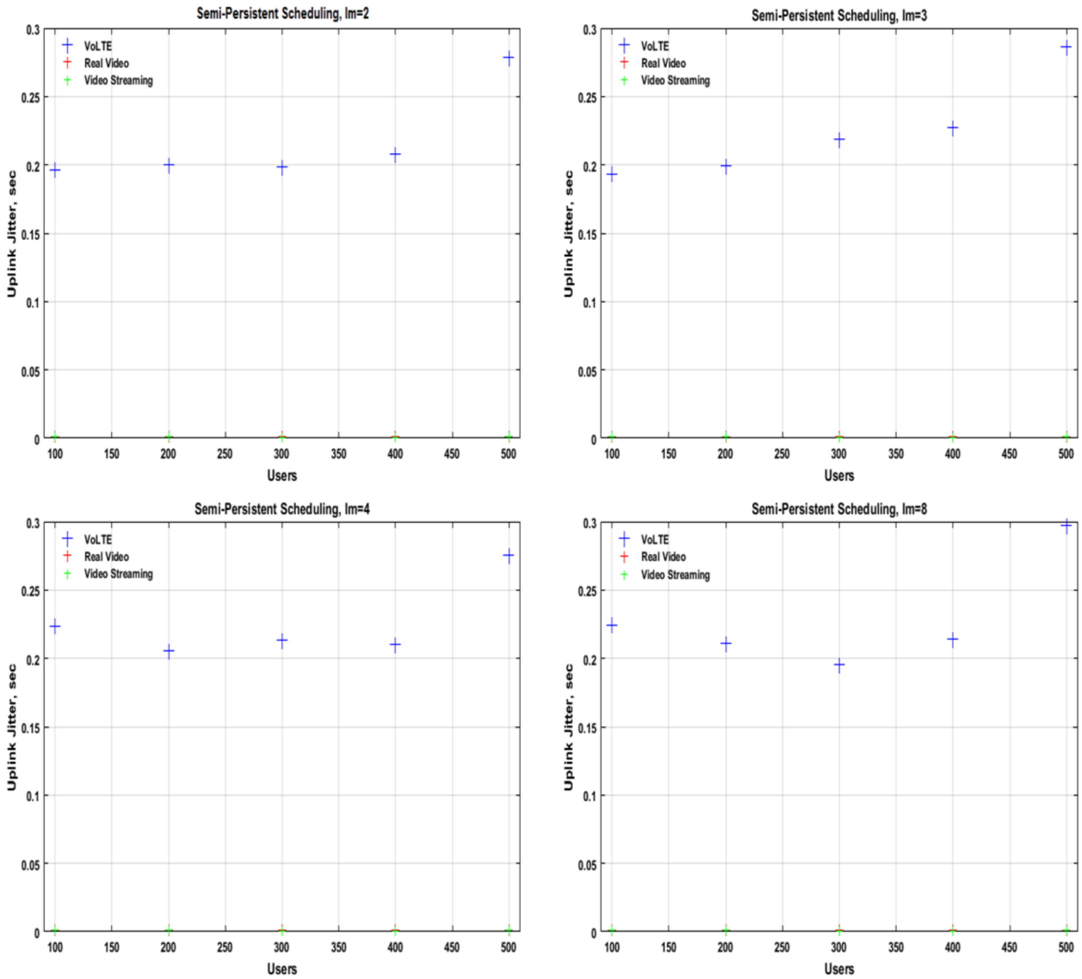



*Figure 61.* Semi Persistent Scheduling Algorithm Jitter per Traffic

In Figure 62 we are able to inspect the performance of the proposed scheme in terms of jitter.

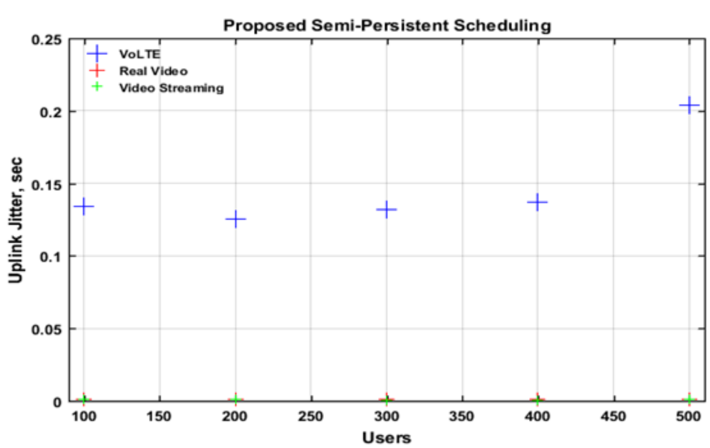



*Figure 62.* Proposed Semi Persistent Scheduling Jitter per Traffic

### 7.2.3 Throughput

The following figures illustrate the throughput of each scheduling method in relation to the number of users in the network. The title of each figure specifies the scheduling technique that was applied. Moreover, in the case of Semi-Persistent Scheduling a variable named "Im" is defined, referring to the value of the `implicitReleaseAfter` parameter that was discussed in the chapters above. The last figure represents the proposed SPS algorithm that conforms the value of `implicitReleaseAfter` according to each user's needs.

While throughput might be a decisive parameter in the examination of network efficiency, this is not the case. The reason behind this, is that the traffic types used in this simulation are not that bandwidth consuming as control limited, especially VoLTE. Therefore, our main concern is the number of simultaneous voice calls that can be handled by PDCCH, while secure delay levels are maintained overall.

To start with, FIFO and DS are not able to catch up with SPS in terms of throughput, because the 20% of resources that were assigned to VoLTE limits that possibility. Yet, SPS thanks to the predominant allocation of the VoLTE users is capable of raising higher throughput values.

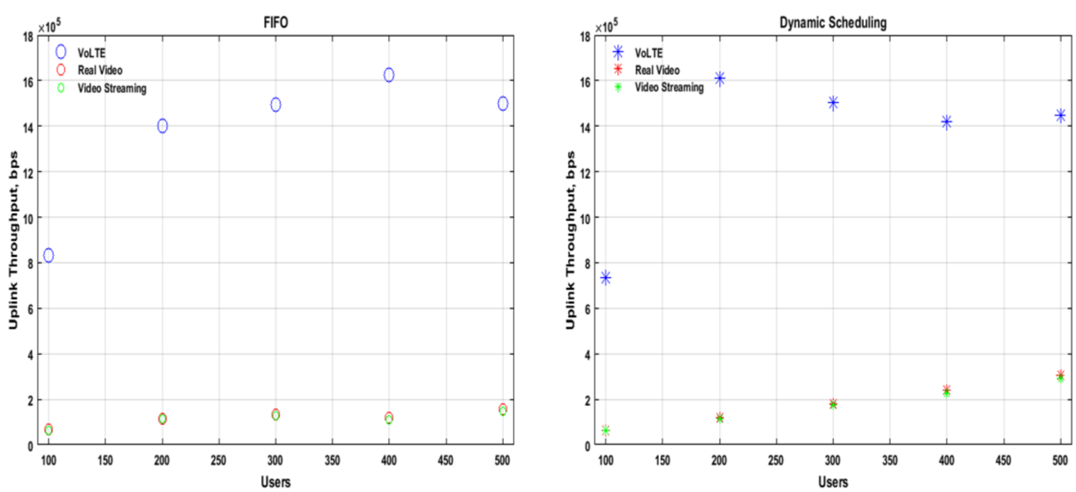



***Figure 63.*** FIFO and Dynamic Scheduling Algorithm Throughput per Traffic

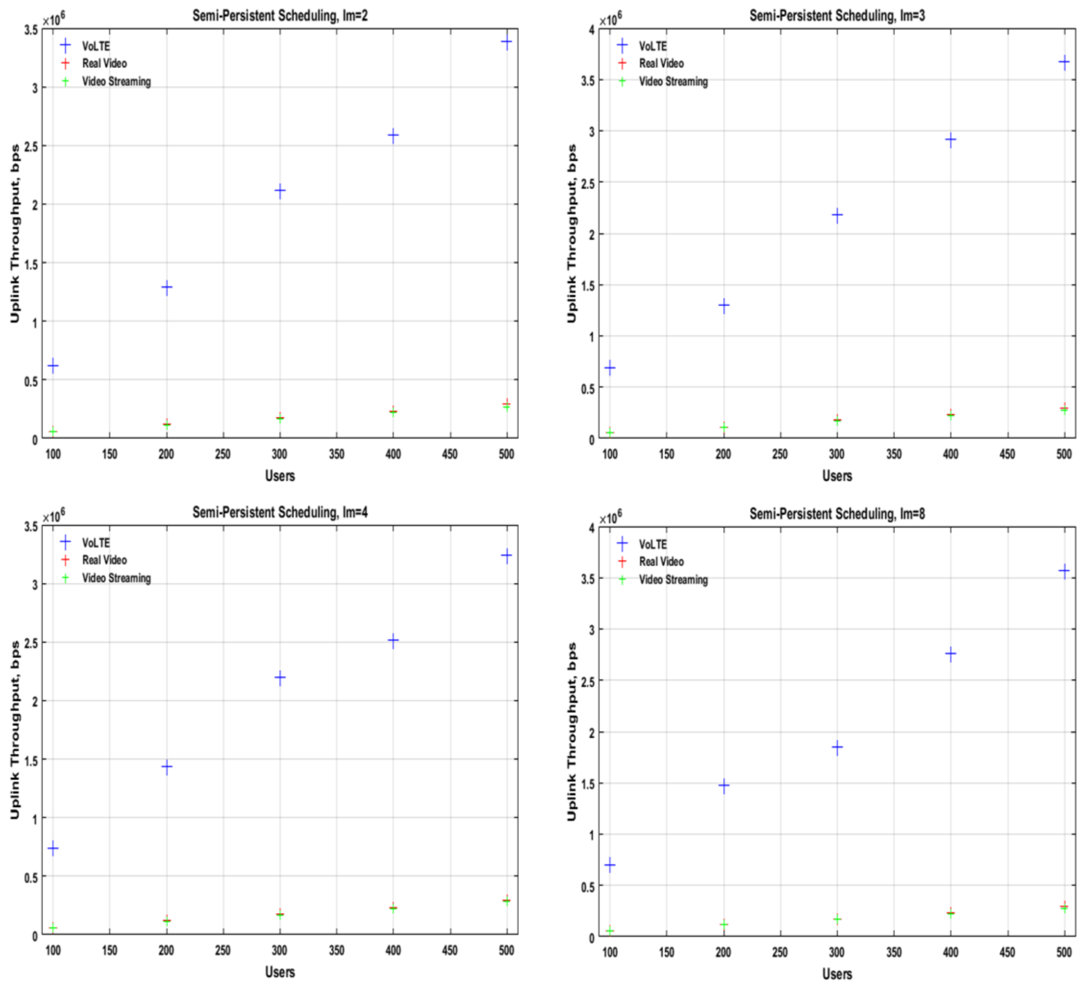



*Figure 64.* Semi Persistent Scheduling Algorithm Throughput per Traffic

In accordance to the aforesaid, the proposed scheme provides similar results to those of SPS, as illustrated in Figure 65.

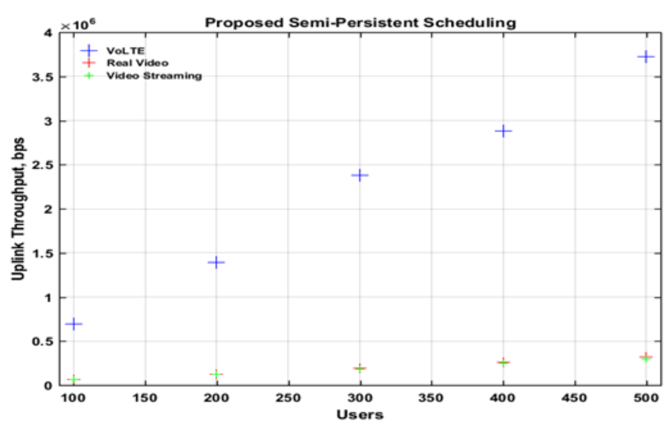



*Figure 65.* Semi Persistent Scheduling Throughput per Traffic

## 7.3 Third Case Scenario (80 10 10)

In the third case scenario we consider that in each downlink and uplink subframe:

- 80% of the resources are available for VoLTE scheduling.
- 10% of the resources are available for Real Video scheduling.
- 10% of the resources are available for Video Streaming scheduling.

However, in the case of Semi-Persistent Scheduling and the proposed edition of that technique, VoLTE traffic is prioritized over the resource allocation scheme discussed above. Thereby, the users that have already initiated a VoLTE call are predominantly allocated so that their QoS need is satisfied.

This case scenario differs from the abovementioned because it provides spare resources for the VoLTE traffic, whilst scarce amounts of assets for the other two traffic types.

### 7.3.1 Delay

Given that we have provided more than enough resources to VoLTE in each subframe, FIFO seems to operate efficiently, even for a considerable number of users. However, as mentioned in the first case scenario, even if we allocated 100% of the resources to VoLTE we cannot avoid the elevation of latency that is observed for 500 or higher numbers of users, because that happens due to PDCCH resource shortage. We are capable of proving that this elevation happens due to control region resource deficit just by taking a look at the first cases scenario delay figures. As we can see, even when the resources that were assigned to VoLTE were 50% of each subframe, the delay increases in the same way.

The differences between the first case scenario and this one arise in the case of Real Video and Video Streaming while using DS. In the first case 25% of the PUSCH resources where enough to support 500 users, while in this case the 10% is not, that's why latency is increased.

Concerning this case scenario, Dynamic Scheduling behaves in the same manner as FIFO, because as soon as PDCCH is able to handle all the VoLTE users in the network congestion is not encountered. On the other hand, SPS retains constant VoLTE delay values and dodges the control channel congestion issue.

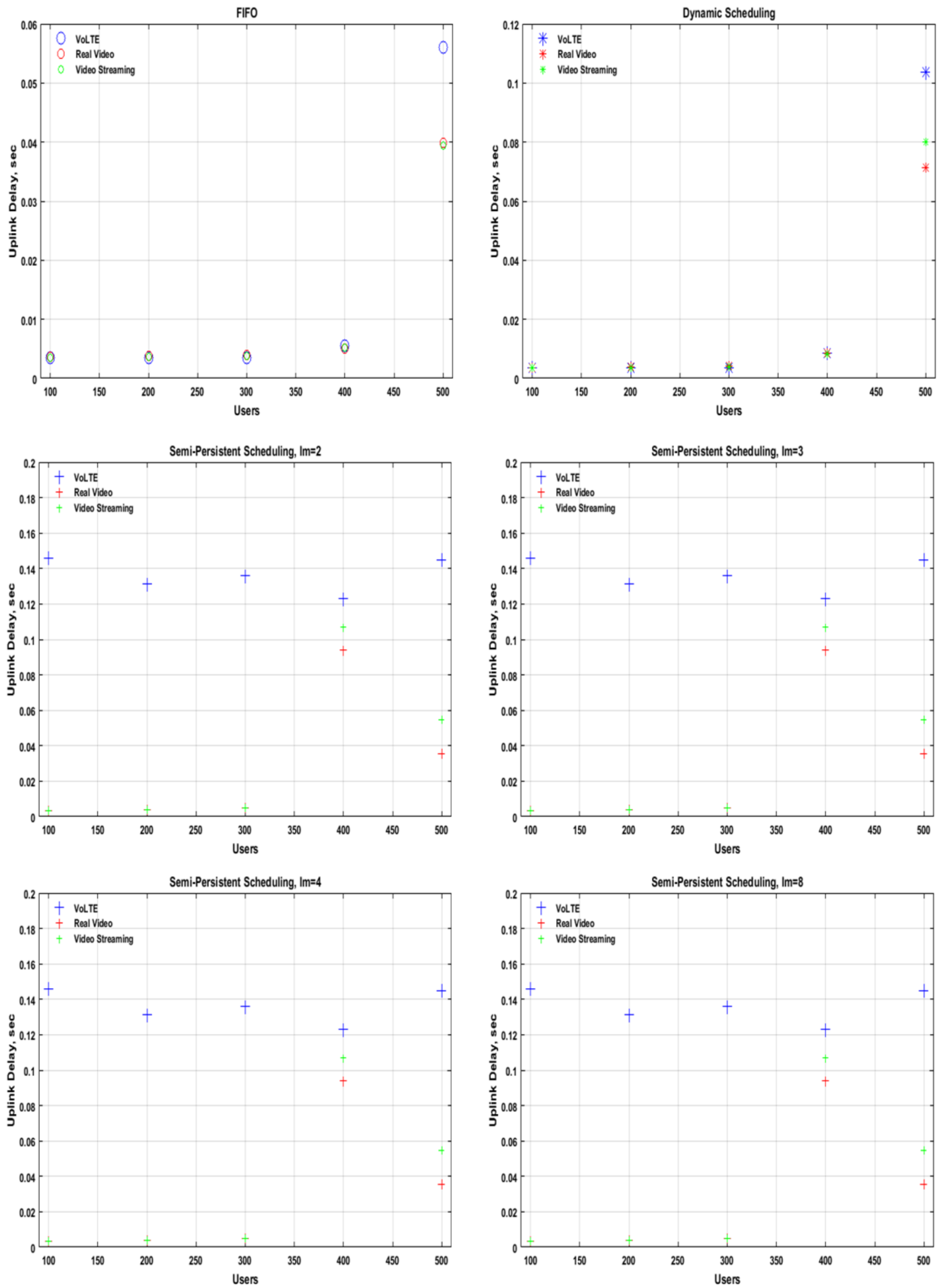



*Figure 66.* FIFO, Dynamic and Semi Persistent Scheduling Algorithm Mean Packet Delay per Traffic

Last but not least, the proposed scheme manages to provide even better mean packet delay values, as shown in Figure 67.

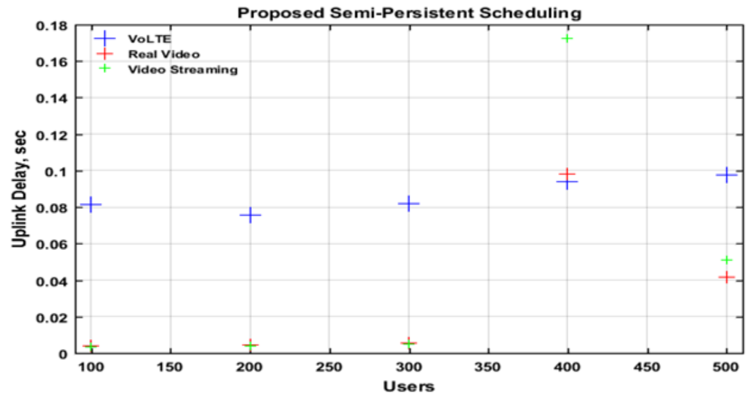



***Figure 67.*** Proposed Semi Persistent Scheduling Mean Packet Delay per Traffic

## 7.3.2 Jitter

The following figures illustrate the jitter of each scheduling method in relation to the number of users in the network. Each figures title specifies the scheduling technique that was applied. Moreover, in the case of Semi-Persistent Scheduling a variable named "Im" is defined, referring to the value of the `implicitReleaseAfter` parameter that was discussed in the chapters above. The last figure represents the proposed SPS algorithm that conforms the value of `implicitReleaseAfter` according to each user's needs.

The behavior of jitter is directly influenced by mean packet delay. Therefore, the above statements about the efficiency of each scheduling method regarding delay are applicable to this case to.

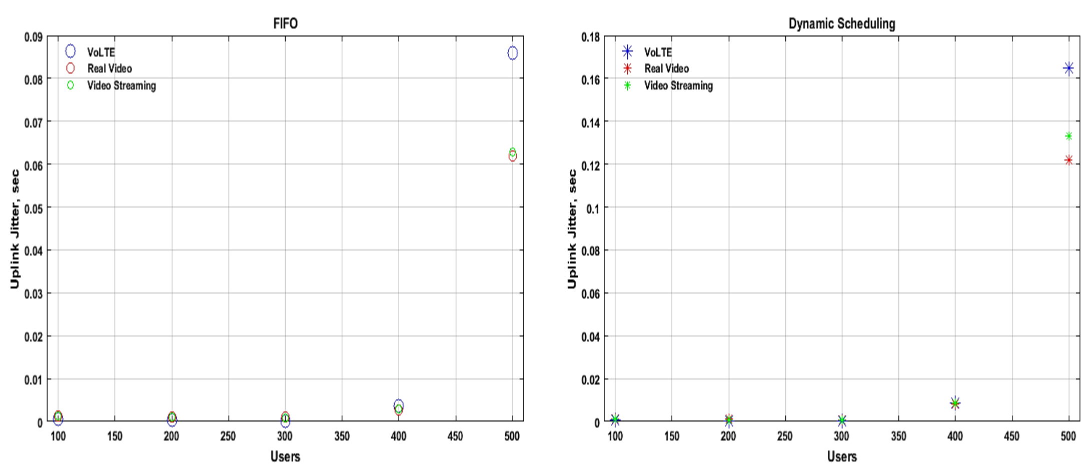



***Figure 68.*** FIFO and Dynamic Scheduling Mean Packet Jitter per Traffic

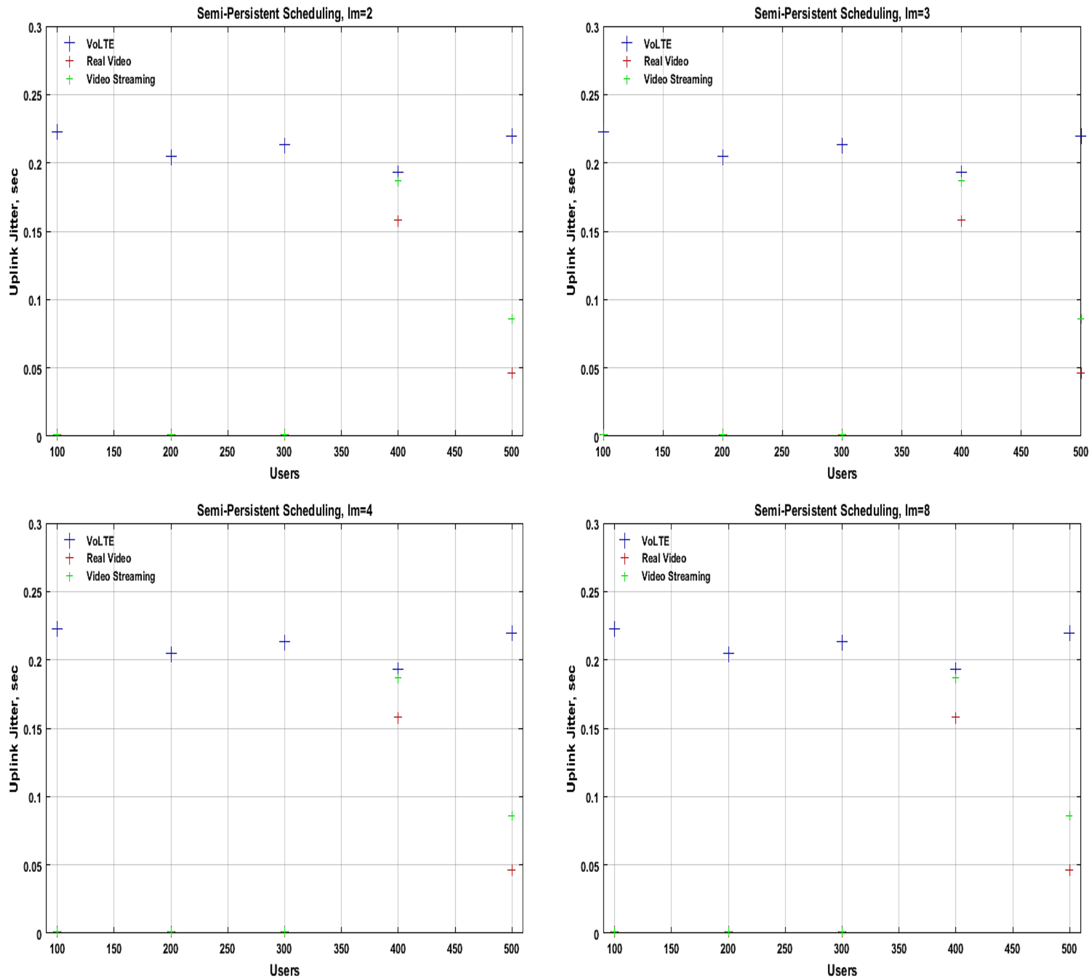



*Figure 69.* Semi Persistent Scheduling Algorithm Jitter per Traffic

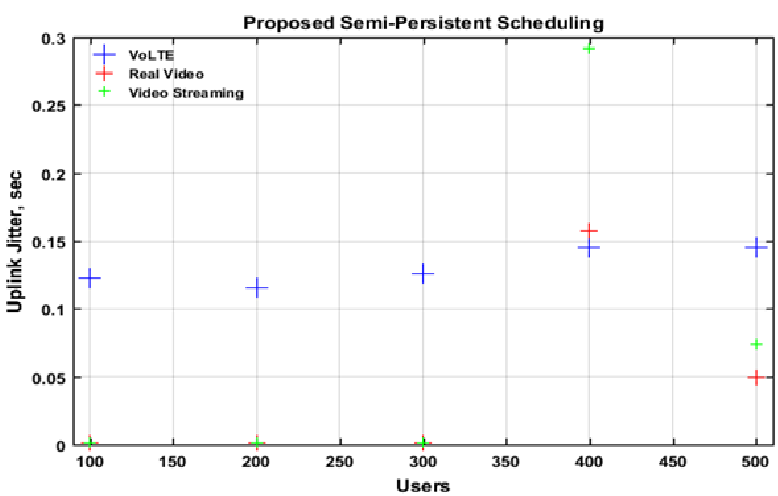



*Figure 70.* Proposed Semi Persistent Scheduling Jitter per Traffic

### 7.3.3 Throughput

The following figures illustrate the throughput of each scheduling method in relation to the number of users in the network. The title of each figure specifies the scheduling technique that was applied. Moreover, in the case of Semi-Persistent Scheduling a variable named "Im" is defined, referring to the value of the `implicitReleaseAfter` parameter that was discussed in the chapters above. The last figure represents the proposed SPS algorithm that conforms the value of `implicitReleaseAfter` according to each user's needs.

While throughput might be a decisive parameter in the examination of network efficiency, this is not the case. The reason behind this, is that the traffic types used in this simulation are not that bandwidth consuming as control limited, especially VoLTE. Therefore, our main concern is the number of simultaneous voice calls that can be handled by PDCCH, while secure delay levels are maintained overall.

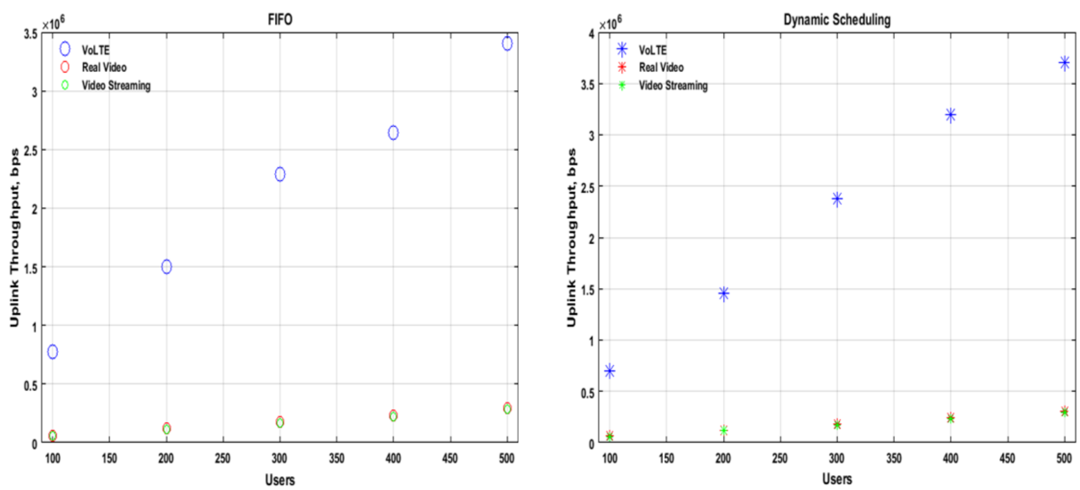



*Figure 71.* FIFO and Dynamic Scheduling Algorithm Throughput per Traffic

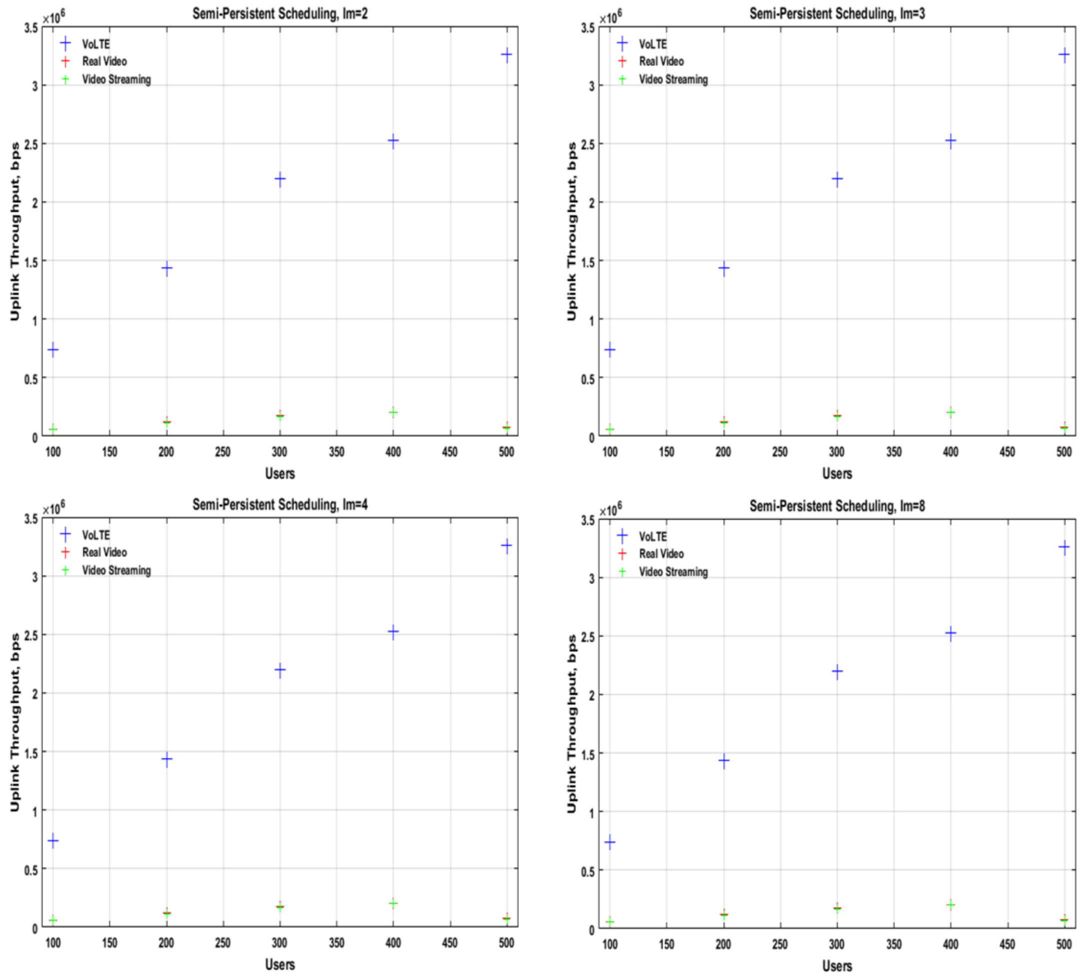



*Figure 72.* Semi Persistent Scheduling Algorithm Throughput per Traffic

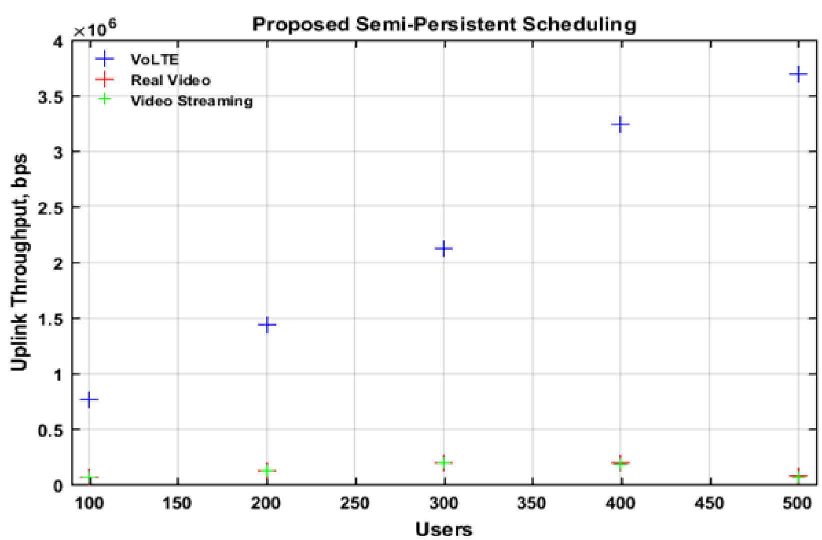



*Figure 73.* Proposed Semi Persistent Scheduling Throughput per Traffic

# 8. Conclusions and Future Work

While this thesis has demonstrated the development of a novel, adaptive and effective scheduling protocol for LTE wireless network, many opportunities for extending its scope remain. Below we summarize some of these directions:

- Given that the duration of each simulation exceeds 24 hours, the transition to a low level programming language is considered crucial.
- The addition of a mechanism that calculates the dropped packets is also considered very important. This mechanism will further enhance the prediction scheme used, providing even better results.
- The inclusion of an algorithm that flunctuates the number of users in the network in real time brings this thesis one step further in the provision of more realistic results.
- The users is this thesis are motionless. Thus, the implementation of an algorithm that handles each user's movement would introduce new features, such as modulation scheme adaptation according to the user's current link conditions.
- The traffic types used in the context of this thesis are limited to VoLTE, Real Video and Video Streaming. Consequently, adding more priority levels supplements the current effort.
- In the current thesis only subframe configuration #1 was applied. Thus, utilizing all seven configurations supplied by LTE provides to the system some sort of flexibility to change the uplink and downlink balance and characteristics. Given that, LTE is able to meet the load conditions.
- The implementation of more machine learning algorithms, such as Support Vector Machines, grants even more reliability regarding the prediction of the `implicitReleaseAfter` value with the ideal outcome.

# Bibliography

[1] T. W. B. Group, "Mobile Cellular Subscriptions (per 100 people)," 2015. [Online]. Available: http://beta.data.worldbank.org.

[2] J. M. Shea, "History of Wireless Communication," 8 12 2011. [Online]. Available: http://wireless.ece.ufl.edu/jshea/.

[3] M. R. Bhalla and A. V. Bhalla, "Generations of Mobile Wireless Technology: A Survey," *International Journal of Computer Applications,* vol. 5, no. 4, pp. 26-32, 2010.

[4] M. Tolstrup, Indoor Radio Planning: A Practical Guide for 2G, 3G and 4G, Wiley; 3 edition, 2015.

[5] E. C. Niehenke, "Wireless Communications: Present and Future," *IEEE Microwave Magazine,* pp. 26-35, 2014.

[6] A. Umar, Mobile Computing and Wireless Communications, NGE Solutions, 2004.

[7] A. Kumar, Y. Liu, J. Sengupta and Divya, "Evolution of Mobile Wireless Communication Networks: 1G to 4G," *International Journal of Electronics and Communication Technology,* vol. 1, no. 1, pp. 68-72, 2010.

[8] A. Sharma, "Generations of Wireless Communication (From 0G to 5G)," [Online]. Available: http://www.academia.edu.

[9] P. Gupta, "EVOLVEMENT OF MOBILE GENERATIONS : 1G To 5G," *International Journal For Technological Research In Engineering,* vol. 1, no. 3, pp. 152-157, 2013.

[10] N. Faltoos, "Generations of Network: 1G, 2G, 3G, 4G, 5G," 2 March 2013. [Online]. Available: http://www.slideshare.net.

[11] P. Sharma, " Evolution of Mobile Wireless Communication Networks-1G to 5G as well as Future Prospective of Next Generation Communication

Network," *International Journal of Computer Science and Mobile Computing,* vol. 2, no. 8, pp. 47-53, 2013.

[12] C. M. Akujuobi and M. N. Sadiku, Introduction to Broadband Communication Systems, Chapman and Hall/CRC, 2007.

[13] "Frame Structure - Downlink," [Online]. Available: http://www.sharetechnote.com.

[14] S. A and E. Mahmoud M., "A Grouped System Architecture for Smart Grids Based AMI Communications over LTE," *International Journal of Wireless & Mobile Networks V,* vol. 7, no. 6, pp. 55-70, 2015.

[15] "Decision Trees," MATLAB, [Online]. Available: http://www.mathworks.com.

[16] "What are the advantages and disadvantages of 3G," [Online]. Available: http://www.careerride.com.

[17] "Difference between 2G and 3G Technology," [Online]. Available: http://www.engineersgarage.com.

[18] I. Poole, "3G HSUPA - High Speed Uplink Packet Access," Radio-Electronics.com, [Online]. Available: http://www.radio-electronics.com.

[19] I. Poole, "HSPA - High Speed Packet Access Tutorial," [Online]. Available: http://www.radio-electronics.com.

[20] J. Wannstrom, "HSPA," [Online]. Available: http://www.3gpp.org.

[21] "WCDMA / HSPA / HSPA+ Technology," Rohde & Schwarz, [Online]. Available: https://www.rohde-schwarz.com.

[22] "An Introduction to 4G Technology," 4Gon Solutions, [Online]. Available: http://www.4gon.co.uk.

[23] C. Cox, An Introduction to LTE: LTE, LTE-Advanced, SAE, VoLTE and 4G Mobile Communications, 2nd Edition, Wiley, 2014.

[24] T. Communications, "LTE Networks: Evolution and Technology Overview," 2010.

[25] "Overview of LTE 3GPP releases," February 2015. [Online]. Available: http://4g-lte-world.blogspot.gr/2015/02/overview-of-lte-3gpp-releases.html.

[26] M. Rumney, LTE and the Evolution to 4G Wireless: Design and Measurement Challenges, 2nd Edition, Wiley, 2013.

[27] C. Johnson, Long Term Evolution IN BULLETS, 2nd Edition, 2012.

[28] 5GNOW, "Deliverables," 3 3 2015. [Online]. Available: http://www.5gnow.eu.

[29] 3GPP, "ETSI TS 36.101 LTE; Evolved Universal Terrestrial Radio Access (E-UTRA); User Equipment (UE) radio transmission and reception".

[30] 3. T. 36.104, "LTE; Evolved Universal Terrestrial Radio Access (E-UTRA); Base Station (BS) radio transmission and reception".

[31] 3. T. 3. 212, "LTE; Evolved Universal Terrestrial Radio Access (E-UTRA); Multiplexing and channel coding".

[32] I. T. Union, From Semaphore to Sattelite.

[33] QUALCOMM, "The Evolution of Mobile Technologies," 2014.

[34] "File:FDMA, TDMA ja CDMA graafiline võrdlus.jpg," 14 12 2013. [Online]. Available: https://commons.wikimedia.org.

[35] K. Rajan, "Dynamic Scheduling, Persistent Scheduling and Semi Persistent Scheduling in LTE," 23 9 2011. [Online]. Available: http://lte-research.blogspot.gr.

[36] Ixia, Long Term Evolution-Access Edition 10, 2013.

[37] A. Banupriya, T. Suba, K. Rajalakshmi and S. Rajasri, "MILESTONE OF WIRELESS COMMUNICATION(1G TO 5G TECHNOLOGY)," *International Journal of Engineering Research and Science & Technology,* vol. 1, no. 1, pp. 39-45, 2015.

[38] M. M. u. i. Mir and D. S. Kumar, "Evolution of Mobile Wireless Technology from 0G to 5G," *International Journal of Computer Science and Information Technologies,* vol. 6, no. 3, pp. 2545-2551, 2015.

[39] S. Mondal, A. Sinha and J. Routh, "A Survey on Evolution of Wireless Generations 0G to 7G," *International Journal of Advance Research in Science and Engineering,* vol. 1, no. 2, pp. 5-10.

[40] H. Zarrinkoub, Understanding LTE with MATLAB, Wiley, 2014.

[41] J. Cao, M. Ma, H. Li, Y. Zhang and Z. Luo, "A Survey on Security Aspects for LTE and LTE-A Networks," *IEEE COMMUNICATIONS SURVEYS & TUTORIALS,* vol. 16, no. 1, pp. 283-302, 2014.

[42] L. Liu, Y. Zhou, L. Tian and J. Shi, "CPC-Based Backward-Compatible Network Access for LTE Cognitive Radio Cellular Networks," *EMERGING APPLICATIONS, SERVICES, AND ENGINEERING FOR COGNITIVE CELLULAR SYSTEMS,* pp. 93-99, 2015.

[43] R. Zhang, Z. Zheng, M. Wang, X. S. Shen and L.-L. Xie, "Equivalent Capacity in Carrier Aggregation-Based LTE-A Systems: A Probabilistic Analysis," *IEEE TRANSACTIONS ON WIRELESS COMMUNICATIONS,* vol. 30, no. 11, pp. 6444-6460, 2014.

[44] N. Saxena, B. J. R. Sahu and Y. S. Han, "Traffic-Aware Energy Optimization in Green LTE Cellular Systems," *IEEE COMMUNICATIONS LETTERS,* vol. 18, no. 1, pp. 38-41, 2014.

[45] S. Rathi, N. Malik, N. Chahal and S. Malik, "Throughput for TDD and FDD 4G LTE Systems," *International Journal of Innovative Technology and Exploring Engineering,* vol. 3, no. 12, pp. 73-77, 2014.

[46] Ericsson, "Ericsson Mobility Report," Douglas Gilstrap, 2013.