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BELAGAVI - 590 018, KARNATAKA**

*Project Report on*

**“Machine Learning Paradigms for the Next Generation Wireless Networks”**

*Submitted in partial fulfillment of the requirements for the degree of*

**BACHELOR OF ENGINEERING**

**In**

**ELECTRONICS AND COMMUNICATION**

**For the academic year 2019-2020**

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## **ACKNOWLEDGEMENT**

The satisfaction and euphoria that accompany the successful completion of any task would be incomplete without the mention of people who made it possible, whose consistent guidance and encouragement crowned our efforts with success.

I consider it as my privilege to express the gratitude to all those who guided in the completion of the internship.

I express my gratitude to Principal, **Dr. Sanjay Jain**, for having provided me the golden opportunity to undertake this internship work in their esteemed organization.

I sincerely thank **Dr. R. Elumalai**, HOD, Department of Electronics and Communication Engineering, CMR Institute of Technology for the immense support given to me.

I express my gratitude to my project guides **Prof. Aritri Debnath Ghosh** (Assistant Professor, Dept. of ECE, CMRIT) for their support, guidance and suggestions throughout the internship work.

Last but not the least, heartfelt thanks to our parents and friends for their support.

Above all, I thank the Lord Almighty for His grace on us to succeed in this endeavor.

## **ABSTRACT**

Using machine learning to address the challenge of assisting in intelligent adaptive learning and decision making in order to meet diverse requirements of next-generation wireless networks.

The different resource allocation schemes for transmissions and re-transmissions depending on the requirements of the underlying service and on the traffic characteristics, focusing on Machine learning.

We provide novel resource allocation schemes for initial transmissions and re-transmissions and derive corresponding analytical models for loss rates. We then show how to set the system parameters that allow meeting the URLLC requirements with low resource consumption.

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## Chapter 1

### INTRODUCTION

The history of Wireless Communications started with the understanding of magnetic and electric properties observed during the early days by the Chinese, Greek and Roman cultures and experiments carried out in the 17th and 18th centuries. Here are some selected events in the development of Wireless Communications, in 1807 French mathematician Jean Fourier discovered Fourier's theorem and in 1820 Danish physicist Hans Christian Orsted discovered the electromagnetic field caused by electric current. The French physicist Dominique Arago showed that a wire became a magnet when current flowed through it. French mathematician and physicist Andre-Marie Ampere discovered electrodynamics and proposed an Electromagnetic Telegraph. In 1831 British scientist Michael Faraday discovered electromagnetic induction and predicted existence of electromagnetic waves. 1864 Scottish mathematician and physicist James Clerk Maxwell formulated the electromagnetic theory of light and developed the general equations of the electromagnetic field. He formulated 20 equations that were later simplified into the 4 basic equations we use today.

Antenna design, the relationship between wave propagation technologies and signal power, are elements that make up a wireless network. Knowledge of science behind the communication provides value when designing a wireless network, allowing understanding of potential complications, such as the signal-to-noise ratio, attenuation and multipath scattering, and channel spacing. One of the principal requirements for wireless communication is that the transmitted EM wave must reach the receiver with ample power to allow the receiver to distinguish the wave from the background noise. Some of the components involved in wireless networking are the antennae, base stations, mobile stations, access points, and wireless PC cards. There is a wide selection of antennae, which can be categorized as omnidirectional and directional, depending on the application and terrain. A major concern in designing wireless networks is the channel spacing—it is a delicate balance between allowing the maximum number of channels and avoiding interference between the channels. Two techniques for maximizing the use of frequencies are frequency reuse and multiple access.

Wireless communication — otherwise known as “over the air” —is the electromagnetic transfer of information between two or more points that are not connected by an electrical conductor. The most common wireless technologies use radio waves. With radio waves, intended distances can be short, such as a few meters for Bluetooth or as far as millions of kilometers for deep-space radio communications. It encompasses various types of fixed, mobile, and portable applications, including two-way radios, cellular telephones, personal digital assistants (PDAs), and wireless networking. Other examples of applications of radio wireless technology include GPS units, garage door openers, wireless computer mouse, keyboards and headsets, headphones, radio receivers, satellite television, broadcast television and cordless telephones. Somewhat less common methods of achieving wireless communications include the use of other electromagnetic wireless technologies, such as light, magnetic, or electric fields or the use of sound.

The term wireless has been used twice in communications history, with slightly different meaning. It was initially used from about 1890 for the first radio transmitting and receiving technology, as in wireless telegraphy, until the new word radio replaced it around 1920. Radios in the UK that were not portable continued to be referred to as wireless sets into the 1960s. The term was revived in the 1980s and 1990s mainly to distinguish digital devices that communicate without wires, such as the examples listed in the previous paragraph, from those that require

wires or cables. This became its primary usage in the 2000s, due to the advent of technologies such as mobile broadband, Wi-Fi and Bluetooth.

Wireless operations permit services, such as mobile and interplanetary communications, that are impossible or impractical to implement with the use of wires. The term is commonly used in the telecommunications industry to refer to telecommunications systems (e.g. radio transmitters and receivers, remote controls, etc.) which use some form of energy (e.g. radio waves, acoustic energy,) to transfer information without the use of wires. Information is transferred in this manner over both short and long distances.

## 1.1 1G

The first commercial 1G mobile network in the world was launched by Nippon Telephone and Telegraph Company (NTT) in Tokyo, Japan on 1 December 1979. The first mobile phones were still car phones, but the network was:

- a cellular network with 88 cell sites with base stations, or radio towers covering all districts of Tokyo (unlike in IMTS where the network was still PSTN);
- handover of the call between different cell sites was supported (unlike in IMTS where the call can only be connected to one radio mast);
- automated switching without the need for human switchboard operator (unlike in ARP where the calls are manually switched).

By 1981, the Nordic countries of Norway and Sweden built their first 1G mobile network based on the Nordic Mobile Telephone (NMT) standard, subsequently Denmark and Finland in 1982. The standard spread in quick succession to Saudi Arabia, Russia, and many other Baltic and Asian countries.

On 13 October 1983, the USA eventually had its commercial cellular network launched by Ameritech in Chicago, based on the AMPS standards. The first hand-held mobile phone invented by Martin Cooper and manufactured by Motorola was introduced at the same time.

In 1984, Malaysia adopted the NMT 450 standard and launched its first cellular network by then Jabatan Telekom (now Telekom Malaysia, privatized in 1987), with mobile phones introduced as ATUR 450.

The early car phones in the Japan NTT network later evolved into “shoulder phones” in 1985 that can be carried on the shoulder like a sling bag.

The European Total Access Cellular System (TACS, later renamed to ETACS) standard was introduced in 1985 and first implemented in the UK. After the first commercial cellular network by NTT, JTACS was later introduced in Japan in 1988.

China, currently boasting the largest mobile subscriber market, launched its first mobile network in 1987 by the Ministry of Posts and Telecommunications of China, using the TACS standard. A nationwide network was completed in the following year.

The First generation of wireless telecommunication technology is known as 1G was introduced in 1980. The main difference between then existing systems and 1G was invent of cellular technology and hence it is also known as First generation of analog cellular telephone. In 1G or First generation of wireless telecommunication technology the network contains many cells (Land area was divided into small sectors, each sector is known as cell, a cell is covered by a radio network with one transceiver) and so same frequency can be reused many times which results in great spectrum usage and thus increased the system capacity i.e. large number of users could be accommodated easily.

Use of cellular system in 1G or First generation of wireless telecommunication technology resulted in great spectrum usage. The First generation of wireless telecommunication technology used analog transmission techniques which were basically used for transmitting voice signals. 1G or first generation of wireless telecommunication technology also consist of various

standards among which most popular were Advance Mobile Phone Service (AMPS), Nordic Mobile Telephone (NMT), Total Access Communication System (TACS). All of the standards in 1G use frequency modulation techniques for voice signals and all the handover decisions were taken at the Base Stations (BS). The spectrum within cell was divided into number of channels and every call is allotted a dedicated pair of channels. Data transmission between the wire part of connection and PSTN (Packet Switched Telephone Network) was done using packet-switched network.

Most of the phones in the 1G era were:

- heavy, most initial models weighing around 3-4 kg;
- for corporate and executive use, not for personal use;
- expensive, the Motorola DynaTac priced at US\$3,995 for example;
- hence a symbol of affluence and social status.

The limitations of 1G mobile technology were:

- Poor sound quality;
- Limited coverage;
- Full analog mode of communication, hence inefficient use of the spectrum;
- Low capacity, FDMA technique does not maximize system capacity;
- Different 1G systems are incompatible with one another, due to different frequency ranges of the systems;
- No roaming supported between different operators;
- Weak security on air interface, no support for encryption;
- No Mobile Assisted Handover and hence more burden on the Mobile Switching Center (MSC).

## 2.2 2G

2G is short for second-generation cellular network. 2G cellular networks were commercially launched on the GSM standard in Finland by Radiolinja (now part of Elisa Oyj) in 1991.[1]

Three primary benefits of 2G networks over their predecessors were that:

1. phone conversations were digitally encrypted.
2. significantly more efficient use of the radio frequency spectrum enabling more users per frequency band.
3. Data services for mobile, starting with SMStext messages.

2G technologies enabled the various networks to provide the services such as text messages, picture messages, and MMS (multimedia messages). All text messages sent over 2G are digitally encrypted, allowing the transfer of data in such a way that only the intended receiver can receive and read it.

After 2G was launched, the previous mobile wireless network systems were retroactively dubbed 1G. While radio signals on 1G networks are analog, radio signals on 2G networks are digital. Both systems use digital signaling to connect the radio towers (which listen to the devices) to the rest of the mobile system.

With General Packet Radio Service (GPRS), 2G offers a theoretical maximum transfer speed of 40 kbit/s.[2] With EDGE (Enhanced Data Rates for GSM Evolution), there is a theoretical maximum transfer speed of 384 kbit/s.

The most common 2G technology was the time division multiple access (TDMA)-based GSM, originally from Europe but used in most of the world outside North America. Over 60 GSM operators were also using CDMA2000 in the 450 MHz frequency band (CDMA450) by 2010.

As opposed to the sporadic and competitive development of 1G cellular networks, the emergence of 2G mobile communication technology took place in a planned and cooperative manner. The

cooperative efforts were pioneered by European countries with the formation and work of:

- R21: As early as 1980, the Telecommunications Commission of CEPT, its Radiocommunications Working Group and its frequency sub-working group R21 managed to have mobile services added at 900 MHz in the International Table of Frequency Allocations, and dedicate a total capacity of 1000 channels (2 x 25 MHz) for new civil mobile use, paving way for the frequency spectrum to be utilized for 2G cellular network;
- GSM: In 1982, the GSM (Groupe Special Mobile) working group was formed by the Telecommunications Commission of CEPT to harmonize the technical and operational characteristics of a public mobile communications system in the 900 MHz band;
- ETSI: Established in 1988 to define the standards and technical specification for GSM (Global System for Mobile communications) technology.

GSM became the predominant 2G technology that swept through most parts of the world, and went on to serve 80% of the mobile market in the decades to come. The USA equivalent is IS-54, also known as Digital-AMPS, which was later replaced by IS-136. In 1991 Radiolinja (now Elisa) launched the first GSM network in Finland. In cases where the 900 MHz frequency range was used for both 1G and 2G systems in Europe, the 1G systems were shut down to make space for the 2G systems.

1992, for the first time, data service was introduced to the mobile network in addition to voice, namely Short Message Service (SMS), which supports a data rate of 9.6kbps. The first SMS was sent by engineer Neil Papworth on December 3rd, 1992 where he typed "Merry Christmas" from a computer to then Vodafone director Richard Jarvis on an Orbitel 901 handset.

GPRS (General Packet Radio Service) also known as 2.5G that supports packet switching for data rates up to 160 kbps was introduced in 1995, and EDGE (Enhanced Data Rates for GSM Evolution) also known as 2.75G that supports 8PSK modulation for data rates up to 500 kbps was introduced in 1997. Nokia 3210 that was launched in 1999 became the third best-selling mobile phone of all time, reaching 150 million units in sales. It is still fondly remembered as many people's first mobile phone that looked sleek and finally without an external antenna. Its replacement model 3310 launched in 2000 was sold 126 million units, and is recognized as an icon from the 2G era so much so that it made a comeback in 2017. Ericsson became a major manufacturer of 2G network equipment, roughly 40% of 2G calls were made through Ericsson equipment in the 1990s.

When 2G was introduced to cellphones, it was praised for several reasons. Its digital signal used less power than analog signals so mobile batteries lasted longer. Environmentally friendly 2G technology made possible the introduction of SMS — the short and incredibly popular text message — along with multimedia messages (MMS) and picture messages. 2G's digital encryption added privacy to data and voice calls. Only the intended recipient of a call or text could receive or read it. However 2G cellphones required powerful digital signals to work, so they were unlikely to work in rural or less populated areas.

### 1.3 3G

3G (short for third generation) is the third generation of wireless mobile telecommunications technology. It is the upgrade for 2.5G and 2.5G GPRS networks, for faster data transfer.[1] This is based on a set of standards used for mobile devices and mobile telecommunications use services and networks that comply with the International Mobile Telecommunications-2000 (IMT-2000) specifications by the International Telecommunication Union. 3G finds application in wireless voice telephony, mobile Internet access, fixed wireless Internet access, video calls and mobile TV.[1]

Third generation mobile phones, or "3G Internet" mobile phones, is a set of standards for wireless mobile communication systems, that promises to deliver quality multimedia services along with high quality voice transmission. Its features are:



- 3G systems comply with the International Mobile Telecommunications-2000 (IMT-2000) specifications by the International Telecommunication Union (ITU).
  - The first 3G services were available in 1998.
  - It provides high speed transmission having data transfer rate more than 0.2Mbps.
  - Global roaming services are available for both voice and data.
  - It offers advanced multimedia access like playing music, viewing videos, television services etc.
  - It provides access to all advanced Internet services, for example surfing webpages with audio and video.
  - It paved the way for the increased usage of smartphones with wide screens as they provided better viewing of mobile webpages, videos and mobile televisions.
- 3G specifications are laid down by two groups, 3GPP and 3GPP2.
- **3GPP (Third Generation Partnership Project) – These specifications are based upon Global System for Mobile (GSM) communications, and are known as Universal Mobile Telecommunications Systems (UMTS). The technologies included in it are –**
    - o Universal Terrestrial Radio Access (UTRA)
    - o General Packet Radio Service (GPRS)
    - o Enhanced Data rates for GSM Evolution (EDGE)
  - **3GPP2 – These specifications are based upon Code Division Multiple Access (CDMA). Two main specifications under this are –**
    - o Wideband CDMA (WCDMA)
    - o CDMA2000

3G telecommunication networks support services that provide an information transfer rate of at least 144 kbit/s. Later 3G releases, often denoted 3.5G and 3.75G, also provide mobile broadband access of several Mbit/s to smartphones and mobile modems in laptop computers. This ensures it can be applied to wireless voice telephony, mobile Internet access, fixed wireless Internet access, video calls and mobile TV technologies.

A new generation of cellular standards has appeared approximately every tenth year since 1G systems were introduced in 1979 and the early to mid-1980s. Each generation is characterized by new frequency bands, higher data rates and non-backward-compatible transmission technology. The first commercial 3G networks were introduced in 2001.

It was 3G technology that paved the way for the smartphones and tablets that we use today to stay connected. 3G provides a stable and relatively fast mobile connection. 3G connection-based networks were introduced in 2001, marking the start of widespread use of the internet on mobile phones. Not long after, smartphones were introduced, offering all of the possibilities of a computer in the palm of your hand.

But what is 3G technology, and how does it work?

3G data technology uses a network of phone towers to pass signals, ensuring a stable and relatively fast connection over long distances. The tower nearest to the user's mobile phone passes data to it.

3G offers speeds that are several times faster or higher than any of its predecessors, including the short-lived 2.5G network which offered internet connectivity. 3G speeds are high enough to allow for audio and video streaming. They've been shown to be perfectly adequate for remote collaboration tools, such as Unified Communications, while offering clear and responsive voice clarity across the line.

But what specific speeds does 3G offer? Keep in mind that specific speeds are determined by the tower networks and individual capabilities of the provider. According to some estimates, 3G

offers a real-world maximum speed of 7.2 Mbps for downloads and 2 Mbps for uploads. While today these numbers may not seem impressive, especially compared to 4G or 5G networks, sometimes 4G technology simply is not available. In these cases, 3G is the default option.

In most instances, 3G is more than capable of providing a stable network connection for tasks including calls, web browsing, and video and audio streaming.

There are at least three reasons by 3G is still relevant today -- even with the wide-spread use of 4G and rapidly emerging use of 5G networks.

For one, 3G remains the most widely used internet on the planet, as it covers a staggering 87% of the populated areas (PDF).

Second, stability is often more important than internet speed. When 4G network connectivity is limited or unreliable, users benefit from using a stable 3G network connection. The stability enables browsing and performing online tasks at an adequate pace instead of trying to maintain a 4G connection.

Third, 3G can make more sense from an efficiency standpoint. 3G tends to use less energy, especially if one is not constantly switching between 3G and 4G due to connectivity fluctuations. If conserving battery life of your device is important - for instance, in an emergency situation - switching a connection to 3G can be optimal. Areas of application:

- Wireless voice telephony
- Fixed wireless Internet access
- Mobile Internet access
- Video calls
- Mobile TV technologies
- Video-on-demand
- Video conferencing
- Tele-medicine
- Global Positioning System (GPS)
- Location-based services

## 1.4 4G

4G is the short name for fourth-generation wireless, the stage of broadband mobile communications that will supercede the third generation (3G). Carriers that use orthogonal frequency-division multiplexing (OFDM) instead of time division multiple access (TDMA) or code division multiple access (CDMA) are increasingly marketing their services as being 4G, even when their data speeds are not as fast as the International Telecommunication Union (ITU) specifies. According to the ITU, a 4G network requires a mobile device to be able to exchange data at 100Mbit/sec. A 3G network, on the other hand, can offer data speeds as slow as 3.84Mbit/sec. From the consumer's point of view, 4G is more a marketing term than a technical specification, but carriers feel justified in using the 4G label because it lets the consumer know that he can expect significantly faster data speeds. 4G is of course faster than 3G and it's the mobile technology that almost everyone is using -- because while 5G is faster still, it's not yet widely available at the time of writing. But there's a whole lot more to 4G than that. So for a rundown of all the benefits, how to get 4G, exactly how fast it is, and what the future holds, read on, because we've got an easily digestible guide to everything you need to know. 4G is the fourth generation of mobile phone technology. It follows on from 3G (third generation) and 2G (second generation) mobile technology. 4G technology builds upon what 3G offers, but does everything at a much faster speed.

Although carriers still differ about whether to build 4G data networks using Long Term Evolution (LTE) or Worldwide Interoperability for Microwave Access WiMAX, all carriers seem to agree that OFDM is one of the chief indicators that a service can be legitimately

marketed as being 4G. OFDM is a type of digital modulation in which a signal is split into several narrowband channels at different frequencies. This is more efficient than TDMA, which divides channels into time slots and has multiple users take turns transmitting bursts or CDMA, which simultaneously transmits multiple signals on the same channel.

When fully implemented, 4G is expected to enable pervasive computing, in which simultaneous connections to multiple high-speed networks will provide seamless handoffs throughout a geographical area. Coverage enhancement technologies such as femtocell and picocell are being developed to address the needs of mobile users in homes, public buildings and offices, which will free up network resources for mobile users who are roaming or who are in more remote service areas. Fourth Generation (4G) mobile phones provides broadband cellular network services and is successor to 3G mobile networks. It provides an all IP based cellular communications. The capabilities provided adhere to IMT-Advanced specifications as laid down by International Telecommunication Union (ITU). Its main features are:

- It provides an all IP packet switched network for transmission of voice, data, signals and multimedia.
- It aims to provide high quality uninterrupted services to any location at any time.
- As laid down in IMT-Advanced specifications, 4G networks should have peak data rates of 100Mbps for highly mobile stations like train, car etc., and 1Gbps for low mobility stations like residence etc.
- It also lays down that 4G networks should make it possible for 1 Gbps downlink over less than 67 MHz bandwidth.
- They provide have smooth handoffs across heterogeneous network areas.

**4G comes in two main categories –**

- **Long – Term Evolution (LTE)– Long – term evolution or LTE is an extension of the 3G technology. It is a standard for high–speed mobile communication, based upon GSM/EDGE and UMTS/HSPA technologies. The peak data rate for download is 100 Mbps and upload is 50 Mbps. The LTE Advanced meets the specifications of IMT–Advanced standard for 4G technology. Its peak data rates are 1000 Mbps for downlink and 500 Mbps for uplink.**
- **Worldwide Interoperability for Microwave Access (WiMAX)– WiMAX is a mobile wireless broadband access (MWBA) standard is sometimes branded 4G. It offers peak data rates of 128 Mbps for downlink and 56 Mbps for uplink over 20 MHz wide channels. The latest version of WiMAX is not compatible to the earlier versions and instead is compatible with LTE.**

With impressive network capabilities, 4G brought the wireless experience to an entirely new level with impressive user applications, such as sophisticated graphical user interfaces, high-end gaming, high-definition video and high-performance imaging. Consumer expectations for mobile handsets and similar products are becoming more and more sophisticated.

Consumers were demanding a better user experience along with more advanced and useful applications on a more ergonomic device.

Although 3G devices were revolutionary for that time, but they will have to improve in areas like imaging and processing power to support future 4G applications like three dimensional (3D) and holographic gaming, 16 megapixel (MPixel) smart cameras and high-definition (HD) camcorders. Applications like these will demand more processing power than the current 3G handsets offer, requiring more efficient applications processors.

The move to 4G networks allowed service providers to offer the impressive applications that will drive users to upgrade to the new phones. Current downlink data rates are less than 10 megabits per second (Mbps); 4G systems offered downlink data rates well over 100 Mbps, an improvement of 10 times over 3G. 4G systems will also have low latency, improving the consumer experience. With flexible network connections, efficient use of spectrum and impressive user applications, 4G offered what consumers wanted.

Long Term Evolution (LTE) technology is sometimes called 3.9G or Super 3G and has been developed by the Third Generation Partnership Project (3GPP) as an improvement to the current Universal Mobile Telecommunications System (UMTS). By using Orthogonal Frequency Division Multiple Access (OFDMA), LTE was able to provide download rates of 150 Mbps for multi-antenna (2×2) multiple-input multiple output (MIMO) for the highest category terminals. For these terminals upload rates in the 50 Mbps range allowed an efficient transfer of data.

LTE makes very efficient use of the available spectrum with channel bandwidths from 1.25 Megahertz (MHz) to 20 MHz. The flexible “slice” will allow LTE to be more easily implemented in countries where 5 MHz is a commonly allocated amount of spectrum. LTE will also co-exist with legacy systems already rolled out around the world.

Latency in a wireless network describes the time it takes between when an action is initiated or requested and when it actually begins. In 3.5G networks, when a phone is in dormant mode and wants to initiate a connection, a several hundred millisecond (ms) delay is common. For transmission of data packets, 50 ms one-way latency is the norm.

Consumers want a connection experience like they get at their homes using a wired broadband connection. LTE will decrease latency to just 50 ms from dormant to connection and a 5 ms one-way latency after that, delivering connection latencies similar to a wired connection.

A new class of mobile devices is emerging that is a convergence of the Smartphone market with the PC market.

These new MID, Mobile Internet Devices, are low-power, high-performance wireless devices, able to deliver a desktop experience on a small footprint, portable device. MIDs deliver an intuitive user interface with touch screens, as well as full featured browser support, high resolution displays, broadband and personal connectivity, a camera, camcorder, navigation, media player, gaming and office productivity applications in a small footprint that can operate all day on a single charge.

The amount of processing performance needed to deliver these new 4G applications will be large. Integrated, multi-core architectures that deliver microprocessors and DSPs on a single chip will be critical to 4G's success.

Products such as TI's OMAP applications processors enable more sophisticated and intuitive user's interfaces and provide a web browsing experience similar to traditional PCs.

To be able to deliver the performance needed for 4G technologies, process technologies must continue towards higher integration. The current 45 nanometer (nm) process in use today allows up to two times the density compared to the previous 65nm process. In addition to cost savings, the 45nm process achieves a 25% performance increase over the 65nm process. Continued integration will increase performance while decreasing costs over time. But all this integration comes at a price, namely the need for sophisticated power management technologies. Shrinking the process technology down to 45nm has an exponential effect on leakage power until it becomes a significant part of a device's total power.

## **1.5 5G**

Fifth-generation wireless (5G) is the latest iteration of cellular technology, engineered to greatly increase the speed and responsiveness of wireless networks. With 5G, data transmitted over wireless broadband connections can travel at multigigabit speeds, with potential peak speeds as

high as 20 gigabits per second (Gbps) by some estimates. These speeds exceed wireline network speeds and offer latency of 1 millisecond (ms) or lower for uses that require real-time feedback. 5G will also enable a sharp increase in the amount of data transmitted over wireless systems due to more available bandwidth and advanced antenna technology.

5G networks and services will be deployed in stages over the next several years to accommodate the increasing reliance on mobile and internet-enabled devices. Overall, 5G is expected to generate a variety of new applications, uses and business cases as the technology is rolled out.

### *How does 5G work?*

Wireless networks are composed of cell sites divided into sectors that send data through radio waves. Fourth-generation (4G) Long-Term Evolution (LTE) wireless technology provides the foundation for 5G. Unlike 4G, which requires large, high-power cell towers to radiate signals over longer distances, 5G wireless signals will be transmitted via large numbers of small cell stations located in places like light poles or building roofs. The use of multiple small cells is necessary because the millimeter wave (MM wave) spectrum-- the band of spectrum between 30 gigahertz and 300 GHz that 5G relies on to generate high speeds -- can only travel over short distances and is subject to interference from weather and physical obstacles, like buildings or trees.

Previous generations of wireless technology have used lower-frequency bands of spectrum. To offset the challenges relating to distance and interference with MM waves, the wireless industry is also considering the use of a lower-frequency spectrum for 5G networks so network operators could use spectrum they already own to build out their new networks. Lower-frequency spectrum reaches greater distances but has lower speed and capacity than MM wave.

The lower frequency wireless spectrum is made up of low- and midband frequencies. Low-band frequencies operate at around 600 to 700 megahertz (MHz), while midband frequencies operate at around 2.5 to 3.5 GHz. This is compared to high-band MM wave signals, which operate at approximately 24 to 39 GHz.

MM wave signals can be easily blocked by objects such as trees, walls and buildings -- meaning that, much of the time, MM waves can only cover about a city block within direct line of sight of a cell site or node. Different approaches have been tackled regarding how to get around this issue. A brute-force approach involves using multiple nodes around each block of a populated area so that a 5G-enabled device can use an air interface -- switching from node to node while maintaining MM wave speeds.

Another approach -- the more feasible one -- for creating a national 5G network is to use a combination of high-, medium- and low-band frequencies. MM wave may be used in densely populated areas, while low- and midband nodes may be used in less dense areas. The low-band frequencies can travel longer and through different objects. One low-band 5G node can stay connected to a 5G-enabled device for up to hundreds of square miles. This means that an implementation of all three bands will give blanket coverage while providing the fastest speeds in the most highly trafficked areas.

### *How fast is 5G?*

5G download speeds can currently reach upwards of 1,000 megabits per second (Mbps) or even up to 2.1 Gbps. To visualize this, a user could start a YouTube video in 1080p quality on a 5G device without it buffering. Downloading an app or an episode of a Netflix show, which may currently take up to a few minutes, can be completed in just a few seconds. Wirelessly streaming video in 4K also becomes much more viable. If on MM wave, these examples would currently need to be within an unobstructed city block away from a 5G node; if not, the download speed would drop back down to 4G.

Low band can stay locked at 5G over longer distances, and even though the overall speed of low-band 5G may be slower than MM wave, low band should still be faster than what would be considered a good 4G connection. Low-band 5G download speeds may be up to 30 to 250 Mbps. Low-band 5G is more likely to be available for more rural locations. Midband 5G download speeds may reach up to 100 to 900 Mbps, and it is likely to be used in major metro areas.

#### Benefits of 5G

Even though the downsides of 5G are clear when considering how easily MM waves can be blocked, or less clear considering radio frequency (RF) exposure limits, 5G still has plenty of worthy benefits, such as the following:

- use of higher frequencies;
- high bandwidth;
- enhanced mobile broadband;
- a lower latency of 1 ms;
- higher data rates, which will enable new technology options over 5G networks, such as 4K streaming or near-real-time streaming of virtual reality (VR); and
- the potential to have a 5G mobile network made up of low-band, midband and MM wave frequencies.

#### *5G vs. 4G: Key differences*

Each generation of cellular technology is separated by not just its data transmission speed, but also a break in encoding methods, which require end users to upgrade their hardware. 4G can support up to 2 Gbps and is slowly continuing to improve in speeds. 4G featured speeds up to 500 times faster than 3G. 5G can be up to 100 times faster than 4G.

One of the main differences between 4G and 5G is the level of latency, of which 5G will have much less. 5G will use orthogonal frequency-division multiplexing (OFDM) encoding, similar to 4G LTE. 4G, however, will use 20 MHz channels, bonded together at 160 MHz. 5G will be up to between 100 and 800 MHz channels, which requires larger blocks of airwaves than 4G.

Samsung is currently researching 6G. Not too much is currently known on how fast 6G would be and how it would operate; however, 6G will probably operate in similar differences in magnitude as between 4G and 5G. Some think 6G may use MM waves on the radio spectrum and may be a decade away.

#### *5G use cases*

5G use cases can range from business and enterprise use to more casual consumer use. Some examples of how 5G can be used include the following:

- streaming high-quality video;
- communication among devices in an internet of things (IoT) environment;
- more accurate location tracking;
- fixed wireless services;
- low-latency communication; and
- better ability for real-time analytics.

In addition to improvements in speed, capacity and latency, 5G offers network management features -- among them network slicing, which enables mobile operators to create multiple virtual networks within a single physical 5G network. This capability will enable wireless network connections to support specific uses or business cases and could be sold on an as-a-service basis. A self-driving car, for example, could require a network slice that offers extremely fast, low-latency connections so a vehicle could navigate in real time. A home appliance, however, could be connected via a lower-power, slower connection because high performance is not crucial. IoT could use secure, data-only connections.



## Chapter 2

### LITERATURE SURVEY

#### ***Ultra Reliable and Low Latency Communications in 5G Downlink: Physical Layer Aspects***

***Hyoungju Jiy , Sunho Parky , Jeongho Yeo , Younsun Kim , Juho Lee , and Byonghyo Shimy  
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Ultra reliable and low latency communications (URLLC) is a new service category in 5G to accommodate emerging services and applications having stringent latency and reliability requirements. In order to support URLLC, there should be both evolutionary and revolutionary changes in the air interface named 5G new radio (NR). In this article, we provide an up-to-date overview of URLLC with an emphasis on the physical layer challenges and solutions in 5G NR downlink. We highlight key requirements of URLLC and then elaborate the physical layer issues and enabling technologies including packet and frame structure, scheduling schemes, and reliability improvement techniques, which have been discussed in the 3GPP Release 15 standardization.

To address diversified services and applications, International Telecommunication Union (ITU) has classified 5G services into three categories: ultra-reliable and low latency communication (URLLC), massive machine-type communication (mMTC), and enhanced mobile broadband (eMBB). To cope with these new service categories, various performance requirements such as massive connectivity, lower latency, higher reliability, and better energy efficiency have been newly introduced. Since the current radio access mechanism cannot support these relentless changes, 3rd Generation Partnership Project (3GPP) introduced a new air interface referred to as New Radio (NR) . The primary goal of NR is to bring entirely new features and technologies that are not necessarily backward compatible with current 4G LTE systems.

URLLC is one of the key services in 5G communications having wide applications including automated controls, tactile internet, remote operations, and intelligent transportation systems. Despite its importance, the physical layer technologies to seamlessly integrate URLLC into 5G NR are in its infancy. In this article, we have discussed the key requirements for URLLC and presented the physical layer enabling technologies. In order to satisfy stringent latency and reliability requirements, many parts of the physical layer should be re-designed, and the techniques presented in this paper can serve as a starting point. Other than the solutions we discussed, there are many interesting issues worth exploring such as the beamforming strategies for control and data part and the reconfigurable URLLC protocol. Also, study of advanced transceiver architecture to support dynamic numerology adaptation and simultaneous decoding is needed. In this article, we put our focus on the URLLC transmission in the downlink, but there are many open issues for the uplink direction such as one-shot access, active user detection, and grant-free transmission.

#### ***Ultra-Reliable Communication in 5G Wireless Systems***

***Popovski, P. (2014). Ultra-Reliable Communication in 5G Wireless Systems. In 1st***



## *International Conference on 5G for Ubiquitous Connectivity*

Wireless 5G systems will not only be “4G, but faster”. One of the novel features discussed in relation to 5G is Ultra-Reliable Communication (URC), an operation mode not present in today’s wireless systems. URC refers to provision of certain level of communication service almost 100 % of the time. Example URC applications include reliable cloud connectivity, critical connections for industrial automation and reliable wire-less coordination among vehicles. This paper puts forward a systematic view on URC in 5G wireless systems. It starts by analyzing the fundamental mechanisms that constitute a wireless connection and concludes that one of the key steps towards enabling URC is revision of the methods for encoding control information (metadata) and data. It introduces the key concept of Reliable Service Composition, where a service is designed to adapt its requirements to the level of reliability that can be attained. The problem of URC is analyzed across two different dimensions. The first dimension is the type of URC problem that is defined based on the time frame used to measure the reliability of the packet transmission. Two types of URC problems are identified: long-term URC (URC-L) and short-term URC (URC-S). The second dimension is represented by the type of reliability impairment that can affect the communication reliability in a given scenario. The main objective of this paper is to create the context for defining and solving the new engineering problems posed by URC in 5G.

Ultra-reliable communication (URC) will be one of the new operating features that will be brought up by the 5G wireless systems. We have provided several motivating scenarios for supporting URC in future wireless applications. We have analyzed the anatomy of a wireless digital link and shown that the introduction of URC requires fundamental rethinking in the relationship between the control information (metadata) and the actual data, since at high reliability levels the way the metadata is encoded and sent cannot be based on the usual “worst case” analysis. The paper introduces the important concept of Reliable Service Composition, where a service is designed to adapt its requirements to the level of reliability that can be attained. For example, a service can have a “minimal variant” that contains messages that can be encoded and transmitted with very high reliability. We have also introduced different types of URC, long- and short-term, respectively, based on the time frame that is used as a reference to determine the latency of the reliable transmission. Finally, we have identified five general types of reliability impairments that need to be carefully modeled if the system is designed to attain ultra-high reliability levels.

## ***Radio Resource Allocation and Retransmission Schemes for URLLC over 5G networks***

***Salah Eddine Elayoubi, Patrick Brown, Matha Deghel, Ana Galindo-Serrano, Salah Elayoubi***

Low latency targets for Ultra-Reliable Low Latency Communications (URLLC) may be conflicting with their stringent reliability requirements due to the need for re-transmissions. We explore in this paper the different resource allocation schemes for transmissions and re-transmissions depending on the requirements of the underlying service and on the traffic characteristics, focusing on Industrial Internet of Things (IIoT). We namely consider schemes with individual reservation versus a pool of contention-based reserved resources. We provide novel resource allocation schemes for initial transmissions and re-transmissions and derive corresponding analytical models for loss rates. We then show how to set the system parameters that allow meeting the URLLC requirements with low resource consumption.

In 5G networks, Ultra-Reliable Low-Latency Communication (URLLC) is the class of services with the most stringent latency and reliability requirements. This class of services is arguably the most challenging and intriguing because, generally speaking, guaranteeing low latency is conflicting with achieving ultra-high reliability. In the 3GPP (3rd Generation Partnership Project) standard, a general URLLC requirement latency is 99.999 % target reliability with 1 ms (two-way) user-plane (Transmission Time Interval (TTI) length is one efficient way to shorten the latency in the system. Another way to reduce latency is to use grant-free scheduling; instead of the Long-Term Evolution (LTE) grant-based scheduling approach, as waiting for the grant penalizes the latency. In this grant-free fast uplink access, neither issuing a scheduling request nor waiting a scheduling grant are required. On the other hand, retransmission is a key enabler for improving the reliability performance, but again, using classical Hybrid Automatic Repeat Request (HARQ) retransmission procedures introduces additional latency and other re-transmission schemes are needed for URLLC.

This paper focuses on the grant-free approach and explores two related access schemes. Specifically, if the packet arrivals are periodic, a cyclic reservation (also known as semi-persistent scheduling) is the most suitable scheme. Under this scheme, each user has preallocated resources that repeat according to a predefined periodicity. If, however, the packet arrivals are sporadic and/or the number of users exceeds the amount of resources, then contention-based access is the appropriate scheme to be exploited. In this case, the users contend to access some shared time and frequency resources which are preallocated for the contention procedures.

For use cases characterized by periodic packet generation, we propose a scheme where sufficient resources are reserved for the packet transmissions of each User Equipment (UE). However, as some of the packets may be lost due to bad radio conditions, retransmissions may be needed and a pool of common resources is also periodically reserved for re-transmissions. If the size of this pool is equal to the amount of resources reserved for first transmissions, all lost packets can be resent. However, we propose to minimize the size of this pool so that the overall resource consumption is reduced while satisfying the target reliability. We also consider a joint optimization of the link level and the resource allocation. Note that this scheme supposes that the latency target allows that at least one acknowledgment (ACK) can be received for retransmissions to occur. If the latency constraint is so tight that no ACK can be received, all users have to retransmit automatically their packets.

For the sporadic packet case, we combine grant-free contention-based scheme with packet repetitions. Indeed, the UE cannot wait for the ACK before retransmitting its (erroneous) packet as in classical HARQ, as the base station may not realize that it attempted a transmission. In this regard, the approach that consists in sending multiple copies of the same packet without waiting

for the acknowledgments was introduced as an efficient way to improve the reliability performance. This approach is already adopted as a solution in the 3GPP standard. Such an approach will result in collisions between some of the (re)transmitted packets, which will impact the reliability level that can be achieved. Hence, it is important to carefully design the contention-based scheme, which will determine the resource allocation policy an active user will follow to send the replicas of each of its packets. In cases where the latency constraint allows for receiving an ACK, we exploit this additional information about the packets that have been correctly received in order to provide a second retransmission opportunity that reduces further the loss rate.

We then derive the minimal amount of resource reservation so that the performance targets are achieved. Even if the two schemes (individual allocation versus contention-based) correspond in general to use cases with different traffic characteristics, there are some use cases where both schemes are possible. For instance, when there is a limited number of users generating sporadic traffic and with a latency budget that allows for receiving an ACK, both schemes can be used and we show the parameter regions where each of the proposed schemes is better.

The original contributions of this paper are the following:

- We provide a general framework for resource allocation for URLLC services in 5G and guidelines on the optimal choices.
- We propose novel resource allocation schemes for transmissions and retransmissions that meet the performance targets with low resource consumption.
- We derive closed form expressions for reliability performance under the different schemes that fit very well with simulation results.
- We propose a cross-layer scheme where both the link level and the resource allocation are considered for meeting the targets.

## ***5G ultra-reliable low-latency communication implementation challenges and operational issues with IoT devices. Electronics***

***by Siddiqi, M.A., Yu, H. and Joung, J., 2019.***

To meet the diverse industrial and market demands, the International Telecommunication Union (ITU) has classified the fifth-generation (5G) into ultra-reliable low latency communications (URLLC), enhanced mobile broadband (eMBB), and massive machine-type communications (mMTC). Researchers conducted studies to achieve the implementation of the mentioned distributions efficiently, within the available spectrum. This paper aims to highlight the importance of URLLC in accordance with the approaching era of technology and industry requirements. While highlighting a few implementation issues of URLLC, concerns for the Internet of things (IoT) devices that depend on the low latency and reliable communications of URLLC are also addressed. In this paper, the recent progress of 3rd Generation Partnership Project (3GPP) standardization and the implementation of URLLC are included. Finally, the research areas that are open for further investigation in URLLC implementation are highlighted, and efficient implementation of URLLC is discussed. The main contribution of the paper is to provide researchers a fast and brief reference to some of the core issues in the implementation of URLLC. Keeping in view the importance of IoT in the coming era, this paper also covers a few most critical aspects of IoT and V2V communication over URLLC. On the basis of issues being covered in this paper, some of the areas that are still open for further investigation in URLLC improvements are also provided to readers. At the end of the article, a possible solution using edge computing is proposed for URLLC implementation. This paper can provide a comprehensive platform for researchers who are looking to study URLLC and its issues with diverse services and applications.

***A Cell Outage Management Framework for Dense Heterogeneous Networks***  
***Oluwakayode Onireti, Member, IEEE, Ahmed Zoha, Member, IEEE, Jessica Moysen, Student Member, IEEE, Ali Imran, Member, IEEE, Lorenza Giupponi, Senior Member, IEEE, Muhammad Ali Imran, Senior Member, IEEE, and Adnan Abu-Dayya, Senior Member, IEEE***

In this paper, we present a novel cell outage management (COM) framework for heterogeneous networks with split control and data planes—a candidate architecture for meeting future capacity, quality-of-service, and energy efficiency demands. In such an architecture, the control and data functionalities are not necessarily handled by the same node. The control base stations (BSs) manage the transmission of control information and user equipment (UE) mobility, whereas the data BSs handle UE data. An implication of this split architecture is that an outage to a BS in one plane has to be compensated by other BSs in the same plane. Our COM framework addresses this challenge by incorporating two distinct cell outage detection (COD) algorithms to cope with the idiosyncrasies of both data and control planes. The COD algorithm for control cells leverages the relatively larger number of UEs in the control cell to gather large-scale minimization-of-drive-test report data and detects an outage by applying machine learning and anomaly detection techniques. To improve outage detection accuracy, we also investigate and compare the performance of two anomaly-detecting algorithms, i.e., k-nearest-neighbor- and local-outlier-factor-based anomaly detectors, within the control COD. On the other hand, for data cell COD, we propose a heuristic Grey-prediction-based approach, which can work with the small number of UE in the data cell, by exploiting the fact that the control BS manages UE-data BS connectivity and by receiving a periodic update of the received signal reference power statistic between the UEs and data BSs in its coverage. The detection accuracy of the heuristic data COD algorithm is further improved by exploiting the Fourier series of the residual error that is inherent to a Grey prediction model. Our COM framework integrates these two COD algorithms with a cell outage compensation (COC) algorithm that can be applied to both planes. Our COC solution utilizes an actor-critic-based reinforcement learning algorithm, which optimizes the capacity and coverage of the identified outage zone in a plane, by adjusting the antenna gain and transmission power of the surrounding BSs in that plane. The simulation results show that the proposed framework can detect both data and control cell outage and compensate for the detected outage in a reliable manner.

In this paper, we have presented a novel COM framework for HetNets with split control and data planes. Two distinct COD algorithms have been proposed, taking into account an expected large number of UEs in the control cells and a small number of UEs in the data cells. For control COD, we have utilized the large-scale data gathering of MDT reports, as recently standardized by 3GPP in Release 10. The solution exploits MDS techniques to reduce the complexity of data processing while retaining pertinent information to develop training models to reliably apply anomaly detection techniques. Furthermore, within the control COD, we have compared the performance of two anomaly-detecting algorithms, i.e., k-NNAD and LOFAD. We found out that k-NNAD, which is a global anomaly detection model, achieved higher detection accuracy of up to 94% compared with LOFAD, which adopts a local approach for classifying abnormal measurement. On the other hand, for data cell outage, we have utilized a heuristic Grey prediction approach, which can reliably work despite of the small number of UEs in the data cells by exploiting the information stemming from the fact that the control BS manages the UE connectivity to the data BS within its coverage. The simulation results have shown that both control and data COD schemes can detect control and data cell outages, respectively, in a reliable manner. In addition, we have proposed an AC-based RL algorithm, which can be applied for both the control and data COC. The proposed COC algorithm solution was based on optimizing the coverage and capacity of the identified outage zone, by adjusting the gains of the antennas

through electrical tilt and downlink transmission power of the surrounding BSs in that plane. Simulation results have shown that the AC-RL algorithm can recover all UEs from outage within a very short time.

## ***Pre-scheduled Resources for Retransmissions in Ultra-Reliable and Low Latency Communications***

***Abreu, R. B., Mogensen, P. E., & Pedersen, K. I. (2017). Pre-scheduled Resources for Retransmissions in Ultra- Reliable and Low Latency Communications. In 2017 IEEE Wireless Communications and Networking Conference (WCNC) IEEE. I E E E Wireless Communications and Networking Conference***

The fifth generation (5G) cellular network demands new solutions to meet, in an efficient way, the stringent targets for ultra-reliable and low latency communication (URLLC), such as 1-10<sup>-5</sup> reliability within 1 ms. In a wireless system, the control signaling of the scheduling process is also a source of errors and delays. Semi-persistent scheduling (SPS) is an option to reduce the signaling, leading to lower latency and improved transmission reliability. However, conventional SPS still applies grant signaling to schedule the retransmission. In this work it is proposed an alternative scheme in which a group of users shares a pre-scheduled resource for retransmission. The benefit is that it provides a retransmission opportunity without needing scheduling control information. Besides that, if the pre-scheduled resource can not be reallocated, the sharing mechanism avoids excessive capacity loss. It is demonstrated through a simple analytical model that, for right grouping sizes and initial transmission error rates, the target error probability e.g. 10<sup>-5</sup> can be achieved. It is also shown that the suggested scheme can provide improved resource efficiency compared to a single conservative transmission which also avoids re-scheduling.

In this paper it was proposed a scheme that employs pre-scheduling of resources shared by a group of URLLC UEs, for retransmissions. The analysis shows that, with the right dimensioning of groups and BLER target, the probability of contention for the shared retransmission can be sufficiently low. This means that the final error probability can be achieved without re-scheduling procedures. The resource efficiency of the method was compared against a single conservative transmission aiming at 10<sup>-5</sup> of error probability. Considering that the reserved resources are wasted when all URLLC UEs initially succeed, it can be seen that the efficiency gain is higher (up to 28% for 256-bit packet) when more UEs

## ***The research of regression model in machine learning field***

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The paper herein will analyze the sale of iced products affected by variation of temperature. Firstly, we will collect the data of the forecast temperature last year and the sale of iced products and then conduct data compilation and cleansing. Finally, we will set up the mathematical regression analysis model based on the cleansed data by means of data mining theory. Regression analysis refers to the method of studying the relationship between independent variable and dependent variable. Linear regression model that corresponds to the practical situation is proposed in the paper, which is to set up simple linear regression model based on practical problem and then to implement the following with the help of the latest and most popular Python3.6. Python3.6 boasts the features of pure object-oriented, platform independence and concise and elegant language. So we will call the corresponding library function to predict the sale of iced products according to the variation of temperature, which will provide the foundation for the company to adjust its production each month, or even each week and each day. As a result, the situation of overproduction can be avoided. Moreover, the other situation as the profit will be affected by the lack of production since the rise of temperature will also be avoided. So the regression model also has reference value for the other fields of marketing.

The paper herein introduces the algorithm and model of the field of machine learning. Linear regression model is used to analyze the sale of iced products of a company and the effect of temperature variation on the sale. Firstly, we cleanse the data collected one year ago and analyze data at the same time. Then we choose forecast temperature as the independent variable and the sale of iced products as the dependent variable to establish a simple linear regression model for analysis. We use the object-oriented programming language Python3.6 and introduce the linear regression function. Programming language can also make data analysis in the field of data mining easier. The final result correctly leads the company to adjust the production and sale of iced products flexibly according to the variation of temperature, which definitely provides great commercial value and offers crucial theoretical foundation for the sale of other companies who produce iced products.



## ***Machine Learning from Theory to Algorithms: An Overview***

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The current SMAC (Social, Mobile, Analytic, Cloud) technology trend paves the way to a future in which intelligent machines, networked processes and big data are brought together. This virtual world has generated vast amounts of data which is accelerating the adoption of machine learning solutions & practices. Machine Learning enables computers to imitate and adapt human-like behaviour. Using machine learning, each interaction, each action performed, becomes something the system can learn and use as experience for the next time. This work is an overview of this data analytics method which enables computers to learn and do what comes naturally to humans, i.e. learn from experience. It includes the preliminaries of machine learning, the definition, nomenclature and applications' describing it's what, how and why. The technology roadmap of machine learning is discussed to understand and verify its potential as a market & industry practice. The primary intent of this work is to give insight into why machine learning is the future.

Digitalization and the Internet revolution have led to a mounting volume of structured and unstructured data which needs to be utilized for analytics. Machine learning as a key technology driver encompasses the intelligent power to harness the knowledge from the available data. Moreover, the adoption of machine learning solutions for complex real-life problems by both researchers & practitioners has made this field a dynamic area of research with an active participation across industries & countries. In this paper, a comprehensive review of Machine Learning process and algorithms is presented. The purpose is clearly to understand the role, advantage and scope of Machine learning as a technology-based solution.

***Machine Learning Paradigms for Next-Generation Wireless Networks***  
***Chunxiao Jiang, Haijun Zhang, Yong Ren, Zhu Han, Kwang-Cheng Chen, and Lajos Hanzo***

Next-generation wireless networks are expected to support extremely high data rates and radically new applications, which require a new wireless radio technology paradigm. The challenge is that of assisting the radio in intelligent adaptive learning and decision making, so that the diverse requirements of next-generation wireless networks can be satisfied. Machine learning is one of the most promising artificial intelligence tools, conceived to support smart radio terminals. Future smart 5G mobile terminals are expected to autonomously access the most meritorious spectral bands with the aid of sophisticated spectral efficiency learning and inference, in order to control the transmission power, while relying on energy efficiency learning/inference and simultaneously adjusting the transmission protocols with the aid of quality of service learning/inference. Hence we briefly review the rudimentary concepts of machine learning and propose their employment in the compelling applications of 5G networks, including cognitive radios, massive MIMOs, femto/small cells, heterogeneous networks, smart grid, energy harvesting, device-to-device communications, and so on. Our goal is to assist the readers in refining the motivation, problem formulation, and methodology of powerful machine learning algorithms in the context of future networks in order to tap into hitherto unexplored applications and services.

Machine learning relies on two phases, the training phase and the testing phase, where the training phase imposes a much higher complexity than the testing phase. Due to the energy constraints and computational complexity constraints of mobile terminals, it is recommended to only implement the testing phase on shirt-pocket-sized mobile terminals.

This article reviewed the benefits of artificial intelligence aided wireless systems equipped with machine learning. We introduced the major families of machine learning algorithms and discussed their applications in the context of next-generation networks, including massive MIMOs, the smart grid, cognitive radios, heterogeneous networks, femto/small cells, D2D networks, and so on. The classes of supervised, unsupervised, and reinforcement learning tools were investigated, along with the corresponding modeling methodology and possible future applications in 5G networks. In a nutshell, machine learning is an exiting area for artificial intelligence aided networking research

## ***A Machine Learning Approach to TCP Throughput Prediction***

***Mariyam Mirza, Joel Sommers, Paul Barford, Xiaojin Zhu***

TCP throughput prediction is an important capability for networks where multiple paths exist between data senders and receivers. In this paper we describe a new, lightweight method for TCP throughput prediction. Our predictor uses Support Vector Regression; prediction is based on both prior file transfer history and measurements of simple path properties. We evaluate our predictor in a laboratory setting where ground truth can be measured with perfect accuracy. We report the performance of our predictor for oracular and practical measurements of path properties over a wide range of traffic conditions and transfer sizes. For bulk transfers in heavy traffic using oracular measurements, TCP throughput is predicted within 10% of the actual value 87% of the time, representing nearly a 3- fold improvement in accuracy over prior history-based methods. For practical measurements of path properties, predictions can be made within 10% of the actual value nearly 50% of the time, approximately a 60% improvement over history-based methods, and with much lower measurement traffic overhead. We implement our predictor in a tool called PathPerf, test it in the wide area, and show that PathPerf predicts TCP throughput accurately over diverse wide area paths.

In this paper, since we have only considered predictors trained and tested on the same path, a number of features, such as the identity of a path and MTU, are implicit. This makes SVR simpler to use for path-specific prediction, because a lot of path details that would be required by other approaches, such as formula-based prediction, can safely be hidden from the user. If two paths are similar in all features, implicit and explicit, then a predictor trained on one path using only Q and L (and possibly AB) can be used directly for the other path. If paths differ in implicit features, then those features will have to be added explicitly to the predictor to allow the predictor to be portable. Fortunately, SVR extends naturally to include new features. If a complete set of features effecting TCP throughput can be compiled, and there is a wide enough set of consistent training data available, then a universal SVR predictor that works on just about any wireline Internet path might be a possibility. A potential challenge here is identifying and measuring all factors – path properties, TCP flavors, operating system parameters – that might affect throughput. However, this challenge can be surmounted to quite an extent by active path measurements, using TCP header fields, and parsing operating systems configuration files to include operating parameters as SVR features.

## ***Machine Learning in Wireless Sensor Networks: Algorithms, Strategies, and Applications***

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Wireless sensor networks monitor dynamic environments that change rapidly over time. This dynamic behavior is either caused by external factors or initiated by the system designers themselves. To adapt to such conditions, sensor networks often adopt machine learning techniques to eliminate the need for unnecessary redesign. Machine learning also inspires many practical solutions that maximize resource utilization and prolong the lifespan of the network. In this paper, we present an extensive literature review over the period 2002-2013 of machine learning methods that were used to address common issues in wireless sensor networks (WSNs). The advantages and disadvantages of each proposed algorithm are evaluated against the corresponding problem. We also provide a comparative guide to aid WSN designers in developing suitable machine learning solutions for their specific application challenges.

Wireless sensor networks are different from traditional networks in various aspects, thereby necessitating protocols and tools that address unique challenges and limitations. As a consequence, wireless sensor networks require innovative solutions for energy aware and real-time routing, security, scheduling, localization, node clustering, data aggregation, fault detection and data integrity. Machine learning provides a collection of techniques to enhance the ability of wireless sensor networks to adapt to the dynamic behavior of its surrounding environment. Table VIII summarizes studies that have adopted machine learning methods to address these challenges from distinct research areas.

From the discussion so far, it became clear that many design challenges in wireless sensor networks have been resolved using several machine learning methods. In this paper, an extensive literature review over the period 2002-2013 on such studies was presented. In summary, adopting machine learning algorithms in wireless sensor networks has to consider the limited resources of the network, as well as the diversity of learning themes and patterns that will suit the problem at hand. Moreover, numerous issues are still open and need further research efforts such as developing lightweight and distributed message passing techniques, online learning algorithms, hierarchical clustering patterns and adopting machine learning in resource management problems of wireless sensor networks.

***Machine Learning for Wireless Networks with Artificial Intelligence: A Tutorial on Neural Networks***

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Next-generation wireless networks must be able to support ultra-reliable, low-latency communication and intelligently manage the internet of things (IoT) devices in a real-time dynamic environment. Such communication requirements and mobile edge and core intelligence can only be realized by integrating fundamental notions of artificial intelligence (AI) and machine learning across the wireless infrastructure and end-user devices. In this context, this tutorial introduces the use of comprehensive concepts of machine learning, in general, and artificial neural networks (ANNs), in particular, and their potential applications in wireless communications. For this purpose, we present a comprehensive overview on a number of key types of neural networks that include feed-forward, recurrent, spiking, and deep neural networks. For each type of neural network, we present the basic architecture and training procedure, as well as the associated challenges and opportunities. Then, we provide a panoramic overview on the variety of wireless communication problems that can be addressed using ANNs. For each application, we present the main motivation for using ANNs along with their associated challenges while also providing a detailed example for a use case scenario. Meanwhile, for each individual application, we present a broad overview on future works that can be addressed using ANNs. In a nutshell, this article constitutes a comprehensive overview of machine learning tailored to the demands of communications and network engineers.

In this paper, we have provided one of the first comprehensive tutorials on the use of machine learning, in general, and neural networks, in particular, for enabling a variety of applications in tomorrow's wireless networks. In particular, we have presented a comprehensive overview on a number of key types of neural networks, that include feed-forward, recurrent, spiking, and deep neural networks. For each type, we have overviewed the basic architecture and training procedure, as well as the associated challenges and opportunities. Then, we have provided a panoramic overview on the variety of wireless communication problems that can be addressed using ANNs. In particular, we have investigated many emerging applications that include unmanned aerial vehicles, wireless virtual reality, mobile edge caching and computing, Internet of Things, and spectrum management. For each application, we have provided the main motivation for using ANNs along with their associated challenges while also providing a detailed example for a use case scenario. Last, but not least, for each individual application, we have provided a broad overview on future works that can be addressed using ANNs.

## Chapter 3

### METHODOLOGY

#### 3.1 URLLC

The fifth-generation cellular mobile networks are expected to support mission critical ultra-reliable low latency communication (URLLC) services in addition to the enhanced mobile broadband applications. The fifth-generation cellular mobile networks are expected to support mission critical ultra-reliable low latency communication (URLLC) services in addition to the enhanced mobile broadband applications.

##### URLLC Specifications

In order to address this wide field of URLLC services in a generic way, two representative 5G requirements have been defined:

- ITU and 3GPP require 5G to be capable of successfully transmitting a 32-byte message over the 5G radio interface within 1 ms with a  $1 - 10^{-5}$  success probability.
- 3GPP further requires 5G to be able to achieve a latency over the 5G radio interface of 0.5 ms that can be provided on average for multiple data transmissions (the fulfillment of this requirement is not needed for the 5G evaluation at ITU).

It should be noted that the service requirements for critical communication services are typically defined end to end. However, the URLLC requirements specified in 3GPP and ITU apply only to the one-way latency over the 5G radio

network, which constitutes only a fraction of the end-to-end latency budget. An additional latency budget would need to be reserved for the other parts of the communication parts, like the core network and an external network. 5G radio functionality for URLLC should be complemented with low-latency core network design, which could be optimized by, for example, local hosting of application functionality.

The design of a low-latency and high-reliability service involves several components: Integrated frame structure, incredibly fast turnaround, efficient control and data resource sharing, grant-free based uplink transmission, and advanced channel coding schemes. Uplink grant-free structures guarantee a reduction in user equipment (UE) latency transmission through avoiding the middle-man process of acquiring a dedicated scheduling grant.

##### *Robust Transmission Modes for ultra reliable communication*

As described earlier, the general URLLC reliability requirement for one transmission of a packet is  $1 - 10^{-5}$  for 32 bytes with a user plane latency of 1 ms. For the physical layer reliability, this corresponds to a maximum block error rate (BLER) of  $10^{-5}$  (i.e., 0.001 percent) which needs to be achieved at a certain channel quality depending on the deployment in which the URLLC service is operated. For LTE, besides fulfilling the requirement above, the work item description on URLLC also considers use cases with requirements that are less stringent in terms of combinations of reliability and latency. The transmission of a packet can comprise several steps, which can include control signaling and data transmission, for example, for assigning resources or HARQ feedback for retransmissions.

This implies that each individual part of the transmission chain should be reliable enough such that the overall reliability for the entire transmission sequence of the packet is achieved. In the following, we discuss some techniques applicable to both NR and LTE, and aspects of the new

NR design that enable ultra-high-reliability transmission. To achieve ultra-reliable transmissions over a fading radio channel, significant signal-to-noise ratio (SNR) margins are needed. Diversity is a key element for providing ultra-reliable transmission for both NR and LTE while also keeping the fading margins at reasonable levels. Diversity can be exploited, for example, in the time, frequency, and spatial domains. In the time domain, diversity can be achieved by repetitions or feedback-based retransmission when the radio channel has changed its fading (i.e., after the channel coherence time); retransmission within the coherence time still provides repetition or coding gains. It is often not possible to exploit time diversity for ultra-reliable low-latency services if the latency requirements are shorter than the channel coherence time. Diversity in the frequency domain can be exploited within the physical bounds of the channel and available bandwidth by using techniques such as distributed resource mapping or frequency hopping. In the spatial domain, multiple antenna configurations at the transmitter and receiver determine the diversity order. For example, the ITU evaluation configuration suggests that for certain urban macro environments, up to eight transmitter/receiver UE antennas can be considered. With multiple transmit antennas, an open-loop transmit diversity technique such as precoder cycling, which cycles through a set of precoding matrices over the bandwidth of the transmission, can provide spatial diversity as a function of frequency. With multiple receive antennas, different versions of the same signal will be available at the receiver. It is less likely that all of these versions will be in deep fade; therefore, they can be combined to effectively increase the signal-to-interference-plus-noise ratio (SINR). The design of the NR control channel provides flexibility to support different service requirements. To ensure high reliability and wide coverage for the physical DL control channel (PDCCH), NR supports sufficiently low code rate transmission for typical URLLC downlink control information (DCI) sizes. The NR physical UL control channel (PUCCH) supports both short formats of duration 1–2 OFDM symbols and long formats of duration 4–14 OFDM symbols, enabling low-latency features for mini-slot-based transmission and ultra-high reliability for UL control transmission. One relevant example is the two-symbol PUCCH, which also enables frequency hopping and thus increased reliability. In LTE, short PDCCH and short PUCCH have been introduced to support low-latency transmission based on sTTI. Similar reliability enhancements such as support for repetition of data and control, compact DCI format, and high aggregation level can also be considered without significant impact on the existing LTE system. To support very low BLER operation at reasonable SINR levels, some form of robust channel coding is required. In LTE, turbo codes are used for data channels and tail-biting convolutional or Reed-Müller codes for control channels. Turbo codes use two encoders at the transmitter and two decoders at the receiver. With this divide-and-conquer approach, turbo codes outperform all previous error-correction codes. Tail-biting convolutional coding is a technique of trellis termination which avoids the rate loss incurred by zero-tail termination at the expense of a more complex decoder.

Reliability enhancement can be achieved by, for example, extending the existing modulation and coding schemes (MCSs) to support operations at lower code rates. In NR, 3GPP has chosen new channel coding techniques, namely low-density parity check (LDPC) codes and polar codes for data and control channels, respectively. The LDPC codes for NR support two base graphs for the parity check matrix. In information theory, a low-density parity-check (LDPC) code is a linear error correcting code, a method of transmitting a message over a noisy transmission channel. The use of two base graphs provides benefits in terms of performance and implementation complexity, supporting a wide range of code rates and information block sizes. In particular, the LDPC is extended to support a code rate of 1/5 without relying on repetition, unlike in LTE, where repetition is used for code rates below 1/3. This allows higher coding gains at low code rates, which are suitable for use cases requiring high reliability like URLLC.

## *Achievable 5G URLLC Latencies*

We now want to understand what latencies are achievable with different URLLC configurations. Since URLLC is about latencies that can be guaranteed with high reliability (e.g.,  $1 - 10^{-5}$ ), we are interested in the worst case latencies (e.g., with maximum slot alignment times) for each URLLC configuration. Both FDD and TDD carrier configurations are considered. For FDD both LTE evolution as well as NR are investigated; for TDD only NR is investigated, since LTE TDD is not considered for URLLC enhancements. Assumptions on timings are made according to the discussions in 3GPP. The following questions are being investigated:

- How does the NR numerology impact latencies?
- How does the latency change when slots of 14 OFDM symbols are complemented by shorter transmissions of 7, 4, or 2 symbols?
- What benefits does SPS UL transmission provide over scheduling request (SR)-based UL transmission?

The latency for a DL packet transmission with  $k$  retransmissions can be expressed as

$$T_{tot} = T_{align} + T_{tx} + 2T_{proc} + k \cdot (2T_{turn} + 2T_{tx}).$$

The worst case alignment delay  $T_{align}$  corresponds to one (mini-)slot duration for FDD and two (mini-)slot durations for TDD, where we assume alternating UL-DL (mini-)slots in TDD as described earlier. The transmission time  $T_{tx}$  equals the (mini-)slot duration. We further assume one (mini-)slot duration as processing time for each transmitter and receiver, which represents the layer 1 and layer 2 processing for the packet transmission. For each retransmission round, two transmission times  $T_{tx}$  are added, plus two turn-around times  $T_{turn}$ .

## *URLLC applications*

URLLC services have been identified in a variety of fields. A commonality among those services is low latency combined with high reliability. Reliability is specified by the failure probability of packets that are not successfully delivered to the receiver within the latency bound, as they are either erroneous or lost, or arrive too late. In other words, reliability guarantees that messages are successfully delivered, and latency bounds can be met. This has introduced a new performance metric for 5G, since in earlier networks latency was predominantly considered in terms of lowest achievable latency or average latency. Some examples of URLLC services are automation of the smart grid energy distribution, industrial process automation, factory automation, automated intelligent transport systems, remote control of machinery, and tactile Internet services. Most URLLC services introduce real-time control applications. In a smart grid this can be the automation of the energy distribution, including detection and restoration of faults; in factory automation it can be the real-time control of the manufacturing robots; and in intelligent transport systems (ITS) it can be the real-time maneuver coordination among autonomous vehicles and with the transportation infrastructure. Achieving URLLC requirements is quite a challenge for 5G networks and will require massive modifications to the system design of the current telecom infrastructure. Owing to the encouraging results achieved with URLLC, it can play an integral role in the 5G era. Although current user requirements are initially based on high bandwidth, latency and reliability are also expected to play a vital role in real-time applications and mission-critical networks. There are a wide range of fields where URLLC communications can be applied

Medical and Health Care Remote surgery/patient diagnosis: Remote surgery or remote patient's diagnosis might be carried out with the help of a robot. In such cases, the reliability of data transmitted as instruction for robot needs to be ultra-reliable because even a slight latency or delay could be very harmful to the patient.



Media/ Entertainment/ Business: Live reporting of an event, live sports events, online gaming, cloud-based entertainment (VR/AR). With the help of technology, the entire world is shrinking in terms of communications. Users desire to be up to date on world events and entertainment in real-time. Even in terms of business, the delay could make a huge impact on trades carried out in the world. In online gaming, the lag could be very frustrating for gamers.

Transport: Drone-based delivery, remote driving, self-driven cars, traffic management, sub-station management (system synchronization, traffic management) .The importance of reliability and latency is self-explanatory in such projects.

Industrial Automation Control systems: automated assembly lines with robots, machine status reports, process surveillance, power grid management. In order to maximize productivity, industries have moved toward automation. Higher reliability and productivity can be obtained by replacing humans with robots in the manufacturing process.

## 3.2 Resource Allocation

From all URLLC use cases, the most challenging ones arise in the industrial sector (IIoT), where latency requirements are of 1 ms Round Trip Time (RTT) and reliability of 99.99999%. Depending on the type of production or activity developed in the industrial site the communication pattern between machines or controllers and machines may vary. In [9], there are two main low-latency groups of use cases classified according to the communication pattern, i.e., the motion control and the discrete automation. Representative examples of the former use cases are motion control of robots, machine tools, as well as packaging and printing machines. Discrete automation encompasses all types of production that result in discrete products: cars, chocolate bars, etc. In motion control applications, a controller interacts with a large number of sensors and actuators. The controller periodically submits instructions to the devices, which return a response within a cycle time. The messages are typically small, e.g., 56 bytes. The cycle time can be as low as 2 ms, setting stringent end-to-end latency constraints on message forwarding (1 ms). Additional constraints on isochronous message delivery add tight constraints on jitter (1  $\mu$ s), and the communication service has also to be highly available (99.9999%). The message transmission in this type of application will therefore follow a deterministic behaviour, that is why in what follows we will refer to them as periodic traffic. For the discrete automation applications, a large number of sensors distributed over the plant forward measurement data to process controllers on a periodic or event-driven base. This use case requires a high communication service availability (99.99%), an end-to-end latency ranging between 1 ms and 100 ms and data rates rather low since each transaction typically comprises less than 256 bytes. We are interested in this paper in these two families of industrial use cases, and will refer to them in the following using the generic terms of deterministic and sporadic traffics.

### DETERMINISTIC PACKET ARRIVALS

We start by use cases with deterministic packet arrivals. We consider a system with N UEs, indexed by  $i$  and show how the resource allocation can be performed for first transmissions and for retransmissions.

Resource allocation:

In FIG.1, a wireless communication system 100 comprises one or more fixed base infrastructure units 101, 102 forming a network distributed over a geographical region for serving remote units in the time and/or frequency domain. A base unit may also be referred to as an access point, access terminal, base, base station, Node-B, eNode-B or by other terminology used in the art. The one or more base units each comprise one or more transmitters for downlink transmissions 104,105 and one or more receivers for receiving uplink transmissions. The base units are

generally part of a radio access network that includes one or more controllers communicably coupled to one or more corresponding base units. The access network is generally communicably coupled to one or more core networks, which may be coupled to other networks, like the Internet and public Switched telephone networks, among other networks. These and other elements of access and core networks are not illustrated but they are well known generally by those having ordinary skill in the art.

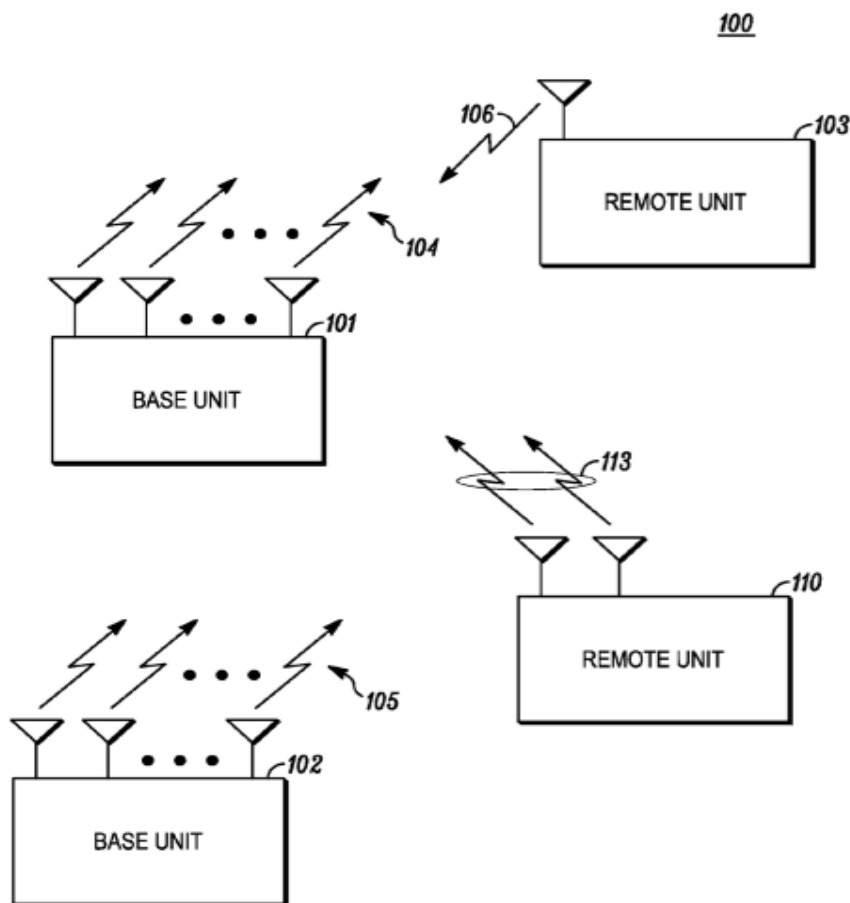


Fig 3.2.1

In FIG. 1, the one or more base units serve a number of remote units 103, 110 within a corresponding serving area, for example, a cell or a cell sector via a wireless communication link. The remote units may be fixed units or mobile terminals. The remote units may also be referred to as sub scriber units, mobiles, mobile stations, users, terminals, Sub scriber stations, user equipment (UE), user terminals, or by other terminology used in the art. The remote units also com prise one or more transmitters and one or more receivers. In FIG. 1, the base unit transmits downlink communication signals to serve remote unit 102 in the time and/or frequency domain. The remote unit 102 communicates directly with base unit 110 via uplink communication signals. A remote unit 108 communicates directly with base unit 112. In one implementation, the wireless communication system is compliant with the Third Generation Partnership Project (3GPP) Universal Mobile Telecommunications System (UMTS) Long Term Evolution (LTE) protocol, also referred to as EUTRA or 3GPP LTE Release-8 (Rel-8) or Some later generation thereof, wherein the base unit transmits using an orthogonal frequency division multiplexing (OFDM) modulation scheme on the downlink and the user terminals transmit on the uplink using a single carrier frequency division multiple access (SC-FDMA) scheme. More

generally, however, the wireless communication system may implement some other open or proprietary communication protocol, for example, WiMAX, among other protocols. The disclosure is not intended to be limited to the implementation of any particular wireless communication system architecture or protocol. In FIG. 2, a user terminal (UE) 200 comprises a controller/processor 210 communicably coupled to memory 212, a data base 214, a transceiver 216, input/output (I/O) device interface 218 connected through a system bus 220. The UE is compliant with the protocol of the wireless communication system within which it operates, for example, the 3GPP LTE Rel-8 or a later generation protocol discussed above. In FIG. 2, the controller/processor 210 may be implemented as any programmed processor. However, the functionality described herein may also be implemented on a general-purpose or a special purpose computer, a programmed microprocessor or microcontroller, peripheral integrated circuit elements, an application-specific integrated circuit or other integrated circuits, hardware/electronic logic circuits, such as a discrete element circuit, a programmable logic device. Such as a programmable logic array, field programmable gate-array, or the like.

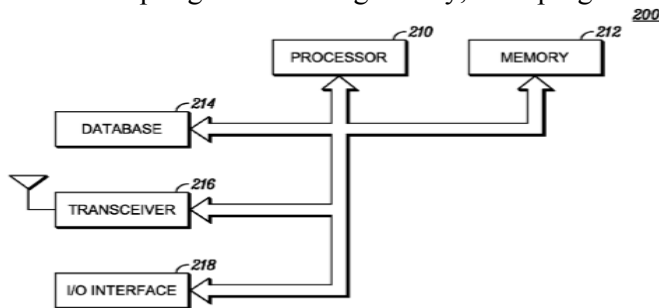


Fig 3.2.2

In FIG. 3.2.2, the memory 212 may include volatile and non-volatile data storage, including one or more electrical, magnetic or optical memories such as a random-access memory (RAM), cache, hard drive, read-only memory (ROM), firmware, or other memory device. The memory may have a cache to speed access to specific data. Data may be stored in the memory or in a separate database. The database interface 214 may be used by the controller/processor to access the database. The transceiver 216 is capable of communicating with user terminals and base stations pursuant to the wireless communication protocol implemented. The I/O device interface 218 connects to one or more input devices that may include a keyboard, mouse, pen-operated touch screen or monitor, Voice-recognition device, or any other device that accepts input. The I/O device interface may also connect to one or more output devices, such as a monitor, printer, disk drive, speakers, or any other device provided to output data. According to one aspect of the disclosure, a wireless communication infrastructure entity, for example, a base station transmits a first control message on an anchor carrier, wherein the first control message includes a resource assignment for the anchor carrier. The base station also transmits a second control message on the anchor carrier. The second control message is associated with a set of component carriers, wherein the set of component carriers are distinct from the anchor carrier. In this regard, the wireless communication infrastructure entity generally comprises a controller that configures a transceiver to transmit a first control message on the anchor carrier and the controller configures the transceiver to transmit a second control message on the anchor carrier wherein the first and second control messages are transmitted such that a resource assignment for the set of component carriers can be determined using the first and second control messages. An anchor carrier is one of the component carriers that a UE is directed to monitor or is capable of monitoring. For example, a Re-8 LTE UE would only be capable of monitoring a Rel-8 compatible LTE carrier but not necessarily a Rel-9 carrier or a carrier Supporting a subsequent release of the LTE specification. In this case the UE monitors the control region (first “n” symbols of each Sub-frame) of its anchor carrier and may not monitor the control region of other

(non-anchor) component carriers. Monitoring includes trying to blindly detect control channels called PDCCH in the control region. In the process flow diagram 300 of FIG. 3, at 310, a wireless communication terminal receives a first control message on an anchor carrier. The first control message includes a resource assignment for the anchor carrier. At 320, the terminal also receives a second control message on the anchor carrier, wherein the second control message is associated with the set of component carriers. The set of component carriers are distinct from the anchor carrier. In one embodiment, the base station transmits the first and second control messages as separate first and second physical downlink control channel (PDCCH) messages. According to this embodiment, the wireless communication terminal receives the first and second control messages as first and second PDCCH messages. More generally, however, the first and second control messages may be some other type of message. In one implementation, the first control message is communicated as a PDCCH message selected from a group of downlink control information DCI formats comprising: 0; 1:1A:1B: 1C:1D; 2 or 2A. Such DCI formats are described in 3GPP TS 36.212, section 5.3.3.1. In one embodiment, the base station is configured to transmit a message identifying the set of component carriers associated with the anchor carrier. Such a message may indicate to the user terminal where resource allocations are expected. In one embodiment, the message identifying the set of component carriers is embedded within a PDCCH message.

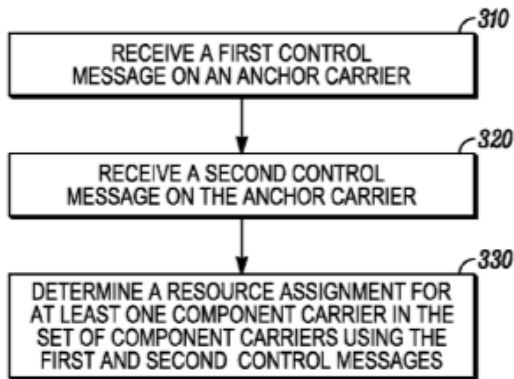


Fig 3.2.3

In FIG. 3.2.3, at 330, the terminal determines a resource assignment for at least one component carrier in the set of component carriers using both the first and the second control messages. More generally, the terminal determines a resource assignment for each component carrier in the set of component carriers using both the first and the second control messages. In one implementation, the resource assignment for at least one component carrier in the set of component carriers is determined using both the first and the second control messages based on bit map information. The bit map information may indicate the presence and/or absence of the resource assignment for the component carrier. In one implementation, the bit map information constitutes part of the first or second control messages. In one implementation, the base station is configured to encode the first control message using a first mask Scrambled with a first identifier and to encode the second control message using a second mask scrambled with a second identifier. In one embodiment, the first and second masks are first and second cyclic redundancy check (CRC) masks and the first and second identifiers are first and second Cell-Radio Network Temporary Identifiers (C-RNTI). The user terminal thus identifies the first control message for the anchor carrier using a first CRC mask scrambled with a first identifier, and the user terminal identifies the second control message is using a second CRC mask scrambled with a second identifier. In an alternate embodiment, the first and second identifiers are predefined identifiers, e.g., integer 0 and 1, respectively, that are combined with the UE Cell-Radio Network Temporary Identifier (C-RNTI). In one implementation, the second control message includes resource assignment type information indicating how to interpret the second control message. In

one embodiment, the resource type information is in the form of offset information. In one implementation, the second control message includes a resource assignment offset for at least one component carrier in the set of component carriers. For example, the offset could be an offset relative to the resource assignment for the anchor carrier. Thus the wireless communication terminal determines the resource assignment for one or more component carriers in the set of component carriers using the resource assignment offset in the second control message. In a more particular implementation, the second control message includes a modulation coding scheme (MCS) offset for at least one component carrier in the set of component carriers. The MCS offset could be an offset relative to the MCS for the anchor carrier or relative to some other reference. According to this implementation, the resource assignment is determined for the one or more component carriers in the set of component carriers using the modulation coding scheme offset in the second control message. In another implementation, the second control message includes information for at least one component carrier wherein the information is selected from a group comprising a hybrid automatic repeat request (HARQ) process identity, a new data indicator, and a redundancy version (RV). The wireless communication terminal then determines the resource assignment for at least one component carrier in the set of component carriers using the information in the second control message. In one embodiment, the functionality performed by the wireless communication terminal during the implementation of the process of FIG. 3 is performed by a processor or controller executing instructions such as program modules, routines, objects, components, data structures stored in memory wherein the processor or controller performs particular tasks or implements corresponding functions. Alternatively, this functionality may be performed by equivalent hardware elements or a combination of hardware and software elements as Suggested above. In a first exemplary implementation, the base station configures the UE via radio resource control (RRC) signalling with an anchor carrier. The UE is expected to only monitor PDCCH messages from the anchor carrier after initial access. Before assigning resources on non-anchor component carriers via individual PDCCH in each component carrier, base unit sends a configuration message to the UE instructing the UE the set of component carriers whose PDCCH messages are also expected to be monitored. The UE is expected to monitor the PDCCH of those component carriers that have been activated by the last received configuration message. The configuration message is signalled to the UE via higher layer signalling, for example a radio resource control (RRC) message. In an alternative embodiment, the configuration message is signalled to the UE via physical layer signalling on a physical downlink shared channel (PDSCH) of the UE's anchor carrier. According to this alternative, a portion of PDSCH resource elements (REs) are punctured to transmit the configuration message. In one embodiment, the information in the configuration message is a long-term bitmap, with each bit corresponding to each component carrier in the set of component carriers that are configured. The UE is expected to monitor the PDCCH of only those component carriers whose bit is set. Alternatively, the information in the configuration message is a single bit. According to this alternative embodiment, the UE is expected to monitor the PDCCH of all the configured component carriers only if the single bit is set. In a second exemplary implementation, the base station configures the UE via radio resource control (RRC) signalling with an anchor carrier. The UE is expected to only monitor PDCCH messages from the anchor carrier after initial access. Before assigning resources on non-anchor component carriers via individual PDCCH messages in each component carrier, the base station or unit sends a configuration message to the UE, instructing the UE the set of component carriers whose PDCCH messages are also expected to be monitored. The UE is expected to monitor the PDCCH messages of those component carriers that have been activated by the last received configuration message. In this implementation, the configuration message is signalled to the UE via physical layer signalling on the physical downlink control channel (PDCCH) of anchor carrier. The configuration message is embedded within an activation PDCCH message (PDCCH-A) and signalled to the UE. The size of the PDCCH-A message can be the same as the existing

3GPP LTE Rel-8 Downlink control information (DCI) format sizes, for example, Format 1C or Format 1A as described in the 3GPP LTE Rel-8 specification TS 36.212, section 5.3.3.1, or a new format with some other size. If the size of PDCCH-A is the same as the existing LTE Rel-8 DCI format sizes, a LTE-A UE can detect PDCCH-A with reduced complexity as the unique number of PDCCH message sizes the UE has to detect is reduced. In some embodiments, the UE identifies the PDCCH-A using a using a LTE Rel-10 specific Cell-Radio Network Temporary Identifier (C-RNTI) assigned to it by the base station. The C-RNTI is used by the base station to scramble the Cyclic Redundancy Check (CRC) bits used for error detection coding of PDCCH-A message. In one embodiment, the information in the configuration message is a long-term bitmap (or a single bit) indicating the component carriers whose PDCCH the UE is expected to monitor. Optionally, an embedded uplink grant can also be included in the configuration message along with the long-term bitmap. The embedded uplink grant instructs the UE to send channel Quality information (CQI) for the set of component carriers identified in the long-term bitmap. If the configuration message is embedded in PDCCH-A, the uplink grant can be made compact and also embedded in PDCCH-A in a manner similar to embedding a Random Access Channel (RACH) response message in a PDCCH as described in the LTE Rel-8 specification (3GPP TS 36.213). The PDCCH-A can also include or indicate resources for acknowledging the transmission of the configuration message to increase the reliability of signalling of the configuration message. Optionally, the base unit can also instruct the UE to send CQI for the set of component carriers identified in the long-term bitmap by signalling a COI-only uplink grant in the same sub-frame where PDCCH-A is transmitted. The configuration message can also optionally include a time offset limit before which the UE should configure its receiver to monitor PDCCH messages from multiple component carriers. The time offset limit could be signalled as a number of sub-frames. Such a time offset limit can help the base unit and the UE identify exactly when the UE will be ready to monitor PDCCH messages from multiple component carriers. In one embodiment to increase the reliability of PDCCH-A reception, the base station assigns resources to the UE on component carriers activated by PDCCH-A only after receiving feedback from the UE. The feedback can be either ACK/ DTX/NACK. ANACK can be signalled by the UE to reject the configuration message from the base unit based on current measurements or based on co-existence optimization. Feedback transmission from the UE is possible on the physical uplink control channel/physical uplink shared channel (PUCCH/PUSCH) of the anchor carrier. The PUCCH resource index implicitly assigned to the UE by the base station based on the lowest index of the Control Channel Element (CCE) on which PDCCH-A is transmitted. When a PDSCH is also scheduled to the UE in the same sub-frame as the PDCCH-A, multiple ACK/NACKs (one each for PDCCH-A and the PDSCH) can be transmitted using multiple PUCCH resources. Alternatively, multiple ACK/ NACKS can be transmitted using ACK/NACK bundling or ACK/NACK channel selection. With a concurrent uplink Physical Uplink Shared Channel (PUSCH) transmission, multiple ACK/NACKs can be accommodated by puncturing the PUSCH transmission. Alternatively, in order to avoid transmission of concurrent ACK/NACKs, a scheduler restriction can be used to not schedule PDSCH in sub-frames on which the PDCCH-A is sent. Optionally, the base station may signal the PDCCH-A more than once to the UE in the same sub-frame to increase reliability of PDCCH-A reception. In a third exemplary implementation, the base station configures the UE via radio resource control (RRC) signalling with an anchor carrier. The UE is expected to only monitor the anchor carrier after initial access. Before assigning resources on component carriers other than the anchor carrier, the base unit sends a configuration message to the UE, instructing it the set of component carriers where PDSCH resource allocations are expected. The configuration message allows the UE to semi-statically configure its receiver to receive the PDSCH on the set of component carriers. The configuration message can be signalled to the UE via RRC signalling. Alternatively, the configuration message can be embedded within an activation PDCCH message (PDCCH-A) and signalled to the UE. In one embodiment, the information in

the configuration message can contain a bitmap (or a single bit) indicating the component carriers on which the PDSCH assignments are expected (long term bitmap). Optionally, an embedded uplink grant can also be included in the configuration message along with the long-term bitmap. The embedded uplink grant for example instructs the UE to send Channel Quality information (CQI) for the set of component carriers identified in the long-term bitmap. Such an uplink grant can be made compact and embedded in the activation PDCCH message in a manner similar to embedding a Random Access Channel (RACH) response message in a PDCCH as described in the LTE Rel-8 specification (3GPP TS 36.213) The PDCCH-A can also include or indicate resources for acknowledging its transmission to increase reliability. Optionally, the base unit can also instruct the UE to send CQI for the set of component carriers identified in the long-term bitmap by signalling a CQI-only uplink grant in the same Sub-frame where PDCCH-A is transmitted. Optionally, the configuration message can also include a time offset limit before which the UE should semi-statically configure its receiver to receive PDSCH on multiple component carriers. The time offset limit could be signalled as a number of Sub-frames. Such a timeoffset limit can help the base unit and the UE identify exactly when the UE will be ready to receive PDSCH on multiple component carriers. To assign resources on multiple component carriers using only the anchor carrier, the base station signals the UE using a first PDCCH message with a Rel-8 compliant DCI format (PDCCH-1) that includes a resource assignment for the anchor carrier and an additional second PDCCH message in the same Sub-frame with multi-component carrier information (PDCCH-2). A Rel-8 compliant DCI format can be one of DCI format 0, 1, 1A, 1B, 1C, 2 or 2A as described in the LTE Rel-8 specification 3GPP TS 36.212, Section 5.3.3.1. While contents of PDCCH-2 messages will be different from Rel-8 PDDCH messages, the size of PDCCH-2 messages can be the same as one of the existing Rel-8 compliant DCI format sizes. This may be beneficial to reduce the number of distinct PDDCH message sizes that a LTE-A UE has to blindly decode. The information in the PDCCH-2 message can be a bitmap indicating the presence or absence of assigned PDSCH resources in each component carrier. The bitmap is valid for only the current sub-frame. Optionally, the information may be a resource allocation offset for each component carrier, or a modulation coding scheme (MCS) offset for each component carrier, or a New Data Indicator (NDI), Redundancy Version indicator (RV) or a Hybrid ARQ process number (HARQID) for each component carrier.

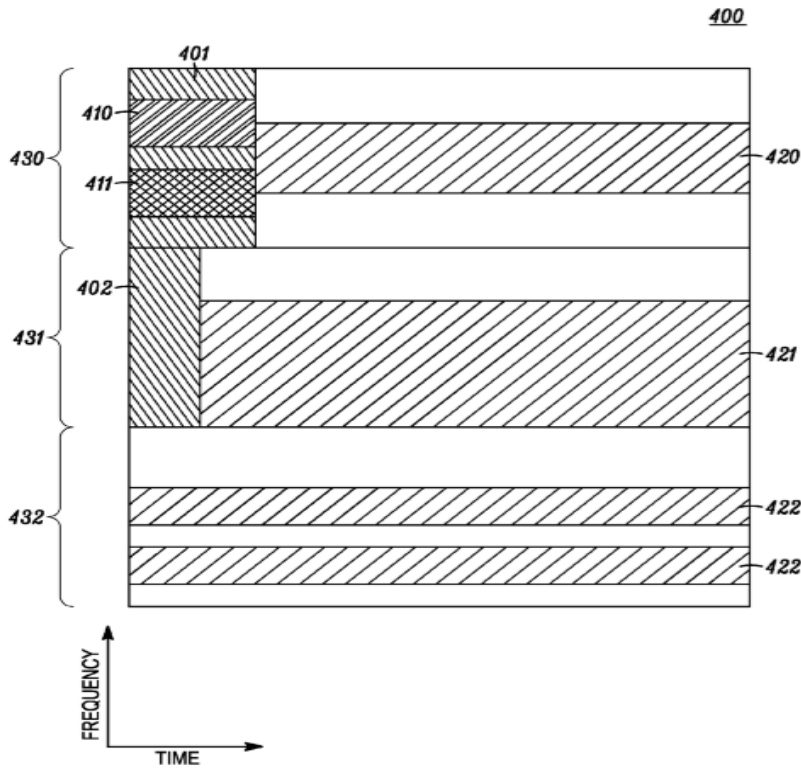


Fig 3.2.4

FIG.3.2. 4 illustrates a Sub-frame 400 in which the PDCCH-1 message 410 and the PDCCH-2 message 411 are signalled. A LTE-A UE detects the PDCCH-1 message 410 in the sub frame by monitoring the PDCCH region 401 of the anchor carrier 430. The UE can identify the PDCCH-1 message 410 using a LTE Rel-8 specific Cell-Radio Network Temporary Identifier (C-RNTI) assigned to it by the base station. The C-RNTI is used to scramble the Cyclic Redundancy Check (CRC) bits used for error detection coding of PDCCH messages. The UE uses Downlink Control Information (DCI) fields in the PDCCH-1 message 410 to determine its PDSCH resource assignment 420 for the anchor carrier 430. The UE further detects PDCCH-2 message 411 also in the PDCCH region 401 of the anchor carrier 430. The UE can identify the PDCCH-2 411 using, for example, a LTE Rel-10 specific C-RNTI assigned to it by the base station. The UE uses DCI information in the PDCCH-2411 and DCI information in the PDCCH-1410 to determine its PDSCH resource assignment 421 for the first component carrier 431 and the PDSCH resource assignment 422 for the second component carrier 432. More generally, the UE can use DCI information in both the PDCCH-1 and the PDCCH-2 messages to determine its PDSCH resource allocations in a set of component carriers that it has been configured to receive. Since both the PDCCH-1 message 410 and the PDCCH-2 message 411 are signalled to the UE on the anchor carrier 430, the UE is not required to monitor the PDCCH region of the component carriers (e.g., PDCCH region 402 of first component carrier 431). This allows the base unit to not configure a PDCCH region in every component carrier thereby reducing control signalling overhead. For example, in FIG. 4 the second component carrier 432 is not configured with a PDDCH region.

Radioresources are allocated into the time/frequency domain. In particular, in the time domain, they are allocated every TTI. In 4G, a TTI lasts for 1 ms, while different TTI sizes are being defined for 5G. In the frequency domain, instead, the total bandwidth is divided in sub-channels whose size depends on the numerology. A combination of a TTI and a subchannel is called Resource Block (RB) and corresponds to the smallest radio resource unit assigned to a UE for data transmission. To guarantee deterministic scheduling, we propose that a periodic resource reservation be performed. In order to satisfy reliability targets



for URLLC, users are assigned a robust Modulation and Coding Scheme (MCS) that ensures a low Block Error Rate (BLER). For a size of an application packet of  $b$  bit, a spectral efficiency of the used MCS of  $\eta$  bit/s/Hz, a bandwidth per RB of  $\omega$  and a TTI  $\tau$ , the number of physical RBs,  $R$ , for transmitting an application packet is  $R = \lceil b/(\eta\tau\omega) \rceil$ , where  $\lceil x \rceil$  (resp.  $\lfloor x \rfloor$ ) denotes the smallest integer larger than  $x$  (resp. the largest integer smaller than  $x$ ). However, some packets will be lost with a packet error rate that depends on the chosen MCS. An additional amount of resources should thus be reserved for retransmissions. This amount is less or equal to the amount of resources reserved initially. As the services are delay-constrained, it is reasonable to allow only one retransmission, but our model can be easily extended to a larger number of retransmissions.

#### Optimal resource allocation

It is worth noting that the latency constraint has a large impact on resource allocation. Indeed, the total time between the packet generation and the termination of its retransmission has to be less than the latency target. This introduces constraints on the amount of TTIs consumed for the transmissions and the retransmissions and thus on the amount of needed spectrum knowing the required amount of resources. Let us now study the impact of this latency constraint on the feasibility of the resource allocation. For the ease of reading and without any loss of generality, we define a "resource unit" (RU) equal to  $R$  RBs, so that each packet occupies 1 unit. Let  $M$  be the amount of RUs per TTI; it is obtained by dividing the amount of available spectrum  $W$  by the available amount of spectral resources per unit:  $M = bW/(R\omega)c$ . The amount of resource units that have to be reserved for first transmissions being equal to the number of UEs  $N$ , the resources for first transmissions are spanned over a number of TTIs equal to  $\lceil N/M \rceil$ . Let the delay before receiving an ACK be equal to  $t_a$  and the delay constraint of the service be equal to  $T$ , the amount of resources allocated to retransmissions,  $K$ , has to verify the following constraint:

$$\lceil K/M \rceil + \lceil N/M \rceil \leq \lceil T - t_a \rceil / \tau$$

The feasibility of this constraint depends on the service and system parameters (latency constraint, ACK response time, number of users, amount of available spectrum). We now suppose that (1) is feasible and derive the optimal value  $K^*$  for satisfying the reliability constraint. We start by observing that the number of lost packets follows a binomial distribution of parameters  $(N, \delta_1)$ , where  $\delta_1$  is the error probability of the first transmissions, as for all  $i \leq N$ , user's  $i$  transmission process is a Bernoulli random variable  $i$  that is equal to 1 with probability  $\delta_1$  and to 0 otherwise. To evaluate the reliability of our resource allocation mechanism, we have to consider two possible events for loss as follows. First, if the number of needed resources for retransmissions is larger than  $K$ , some of the lost packets cannot be retransmitted, leading to a definite loss. Second, even if there is enough space for a retransmission, the retransmission may fail again. Note also that, for the first event, i.e., when there is not enough space to accommodate all retransmissions, we select randomly  $K$  packets among the lost ones for retransmission.

The final error rate for a UE is:

$$e(K, \delta_1, \delta_2) = \sum_{n=0}^{N-1} C_{N-1}^n \delta_1^{n+1} (1 - \delta_1)^{N-1-n} \times \left( \delta_2 I_{n+1 \leq K} + \frac{\delta_2 K + n + 1 - K}{n + 1} I_{n+1 > K} \right)$$

where  $C_n^N$  is the binomial coefficient and  $I_A$  is an indicator function equal to 1 if condition  $A$  is verified and to 0 otherwise.  $\delta_2$  denotes the error rate for a second transmission. Proof. The probability of having  $n$  packets lost in the first round is given by the binomial law as errors are independent. The sum represents the events of having  $n$  lost packets among the first transmissions of the  $N - 1$  other users. The first term within the sum is the binomial law, multiplied by  $\delta$  to consider that the user of interest has lost its packet; and the second term

characterizes the retransmissions. Here, if there is enough space for all lost packets to be retransmitted, the error probability is  $\delta^2$ , giving the term  $\delta^2 I_{n+1} \leq K$ . Otherwise, it is equal to 1 if the packet is not selected for retransmissions (with probability  $n+1-K$ ), giving the term  $n+1-K$   $I_{n+1} > K$  and to  $\delta^2$  if it is selected (giving the term  $\delta^2 K$   $I_{n+1} > K$ ). Note that the error probability reduces to  $\delta^2$  for  $K = N$ .

## Chapter 4

### RESULT

A well-fitting regression model results in predicted values close to the observed data values. The mean model, which uses the mean for every predicted value, generally would be used if there were no informative predictor variables. The fit of a proposed regression model should therefore be better than the fit of the mean model.

Three statistics are used in Ordinary Least Squares (OLS) regression to evaluate model fit: [R-squared](#), the overall F-test, and the Root Mean Square Error (RMSE). All three are based on two sums of squares: Sum of Squares Total (SST) and Sum of Squares Error (SSE). SST measures how far the data are from the mean, and SSE measures how far the data are from the model's predicted values. Different combinations of these two values provide different information about how the regression model compares to the mean model.

R-squared has the useful property that its scale is intuitive: it ranges from zero to one, the same units as the response variable. Lower values of RMSE indicate better fit. RMSE with zero indicating that the proposed model does not improve prediction over the mean model, and one indicating perfect prediction. Improvement in the regression model results in proportional increases in R-squared.

The RMSE is the square root of the variance of the residuals. It indicates the absolute fit of the model to the data—how close the observed data points are to the model's predicted values. Whereas R-squared is a relative measure of fit, RMSE is an absolute measure of fit. As the square root of a variance, RMSE can be interpreted as the standard deviation of the unexplained variance, and has the useful property of being in SE is a good measure of how accurately the model predicts the response, and it is the most important criterion for fit if the main purpose of the model is prediction.

The best measure of model fit depends on the researcher's objectives, and more than one are often useful. The statistics discussed above are applicable to regression models that use OLS estimation. Many types of regression models, however, such as mixed models, generalized linear models, and event history models, use maximum likelihood estimation. These statistics are not available for such models.

After training regression models in Regression Learner, you can compare models based on model statistics, visualize results in response plot, or by plotting actual versus predicted response, and evaluate models using the residual plot. If you use  $k$ -fold cross-validation, then the app computes the model statistics using the observations in the  $k$  validation folds and reports the average values. It makes predictions on the observations in the validation folds and the plots show these predictions. It also computes the residuals on the observations in the validation folds. If you use holdout validation, the app computes the accuracy scores using the observations in the validation fold and makes predictions on these observations. The app uses these predictions in the plots and also computes the residuals based on the predictions. In the response

plot, view the regression model results. After you train a regression model, the response plot displays the predicted response versus record number. If you are using holdout or cross-validation, then these predictions are the predictions on the held-out observations. In other words, each prediction is obtained using a model that was trained without using the corresponding observation. To investigate your results, use the controls on the right. You can:

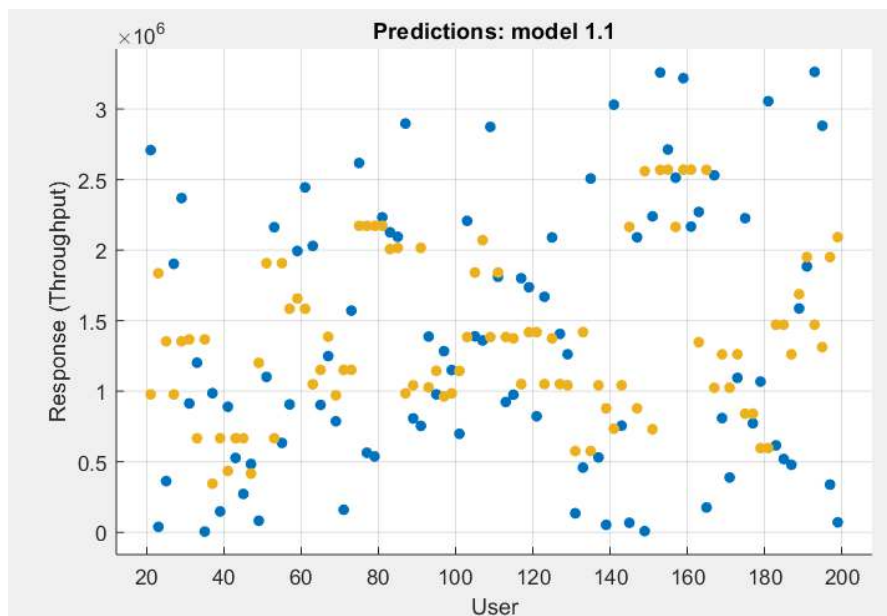
Plot predicted and/or true responses. Use the check boxes under **Plot** to make your selection.

Show prediction errors, drawn as vertical lines between the predicted and true responses, by selecting the **Errors** check box.

Choose the variable to plot on the  $x$ -axis under **X-axis**. You can choose either the record number or one of your predictor variables.

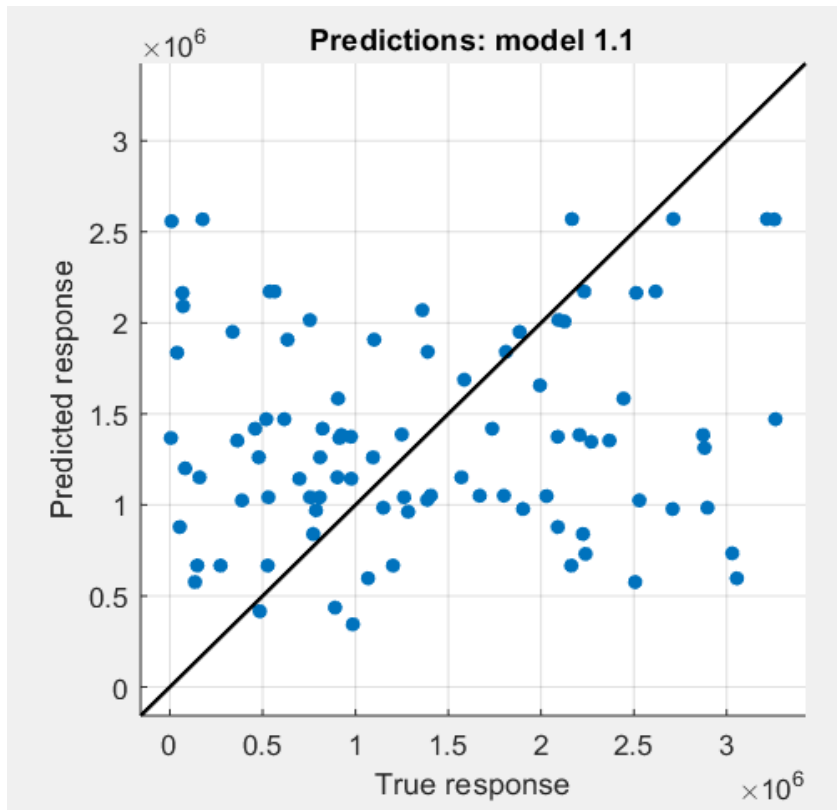
In Regression Learner, use the response plot to try to identify predictors that are useful for predicting the response. To visualize the relation between different predictors and the response, under **X-axis**, select different variables in the **X** list.

Before you train a regression model, the response plot shows the training data. If you have trained a regression model, then the response plot also shows the model predictions.



Use the Predicted vs. Actual plot to check model performance. Use this plot to understand how well the regression model makes predictions for different response values. When you open the plot, the predicted response of your model is plotted against the actual, true response. A perfect regression model has a predicted response equal to the true response, so all the points lie on a diagonal line. The vertical distance from the line to any point is the error of the prediction for that point. A good model has small errors, and so the predictions are scattered near the line. Usually a good model has points scattered roughly symmetrically around the diagonal line. If you can see any clear patterns in the plot, it is likely that you can improve your model. Try training a different model type or making your current model type more flexible using the **Advanced** options in the **Model Type** section. If you are unable to improve your model, it is possible that you need more data, or that you are missing an important predictor.

Scatter plots of Actual vs Predicted are one of the richest form of data visualization. You can tell pretty much everything from it. Ideally, all your points should be close to a regressed diagonal line. So, if the Actual is 5, your predicted should be reasonably close to 5 to. If the Actual is 30, your predicted should also be reasonably close to 30. So, just draw such a diagonal line within your graph and check out where the points lie. If your model had a high R Square, all the points would be close to this diagonal line. The lower the R Square, the weaker the Goodness of fit of your model, the more foggy or dispersed your points are (away from this diagonal line).



Error rate equation:

```
Editor - C:\Users\clera rakshana\Downloads\project\simulation1.m
simulation1.m x p1.m x p2.m x p3.m x Untitled5.m x
1 - N= input('Enter the no of users ');
2 - K=input('enter the value');
3 - n=0:N-1;
4 - d1=1.1;
5 - d2=0.3;
6 - I=rand(5,1);
7 - f1=((nchoosek(n,N-1)))
8 - c=(power(d1,(n+1))).*(power((1-d1),(N-1-n)))
9 - f1=f1.';
10 - c=c.';
11 - f=f1*c;
12 - f=f.';
13 - f3=((d2*I((n+1)<=K)));
14 - c2=(d2*K+n+1-K)/(n+1);
15 - c3=I((n+1)>K);
16 - f4=f3+c2*c3
17 - y=(f*f4)
18 - f5=symsum(y,n,0,N-1)
19
```

```
Command Window
New to MATLAB? See resources for Getting Started.
y =
    2.4360

Undefined function 'symsum' for input arguments of type 'double'.

Error in simulation1 (line 18)
f5=symsum(y,n,0,N-1)
fx
```

```

Editor - C:\Users\clera rakshana\Downloads\project\simulation1.m
simulation1.m x +
1 N= input('Enter the no of users ');
2 K=input('enter the value');
3 d1=0.1;
4 d2=0.3;
5 y=[]
6 for n=1:N
7
8     f1=(nchoosek(N,n))
9     c=(power(d1,(n+1)).*(power((1-d1),(N-1-n)))
10    %f1=f1.';
11    %c=c.';
12    f=f1*c;
13    %f=f.'
14
15    if n+1<=K
16        I1=1;
17    else
18        I1=0;
19    end
20    a1=d2*I1;
21    if n+1>K
22        I2=1;
23    else
24        I2=0;
25    end
26    a2=((d2*K+n+1-K)/(n+1))*I2
27    f2=a1+a2
28    y(n)=f*f2
29
30 end

```

Workspace	
Name ^	Value
a1	0
a2	0.7000
c	1.1111e-07
d1	0.1000
d2	0.3000
f	1.1111e-07
f1	1
f2	0.7000
I1	0
I2	1
K	3
n	6
N	6
y	[0.0118,0.0033,7....

```

>> simulation1
Enter the no of users 6
enter the value3

```

```

y =

    0.0118    0.0033    0.0008    0.0001    0.0000    0.0000

```

## Chapter 5

### CONCLUSION

Ensuring an effective way through which transmissions and subsequent retransmissions of resources take place is crucial for reducing latency. In this work, the error rate minimization mechanism is implemented that reduces the block error rates (BLER) for multiple transmissions and retransmissions. Uplink multiple transmission schemes for 5G Ultra-Reliable Low Latency Communications (URLLC) traffic. The URLLC class of services has been defined for applications requiring extremely stringent latency and reliability. We show that, in systems with episodic traffic and many users compared with the number of transmission resources, randomly transmitting multiple copies of a packet allows to meet the URLLC requirements. We develop analytical models for the packet loss rate for two contention based multiple transmission schemes and show that one dominates the other in the parameter range for which the URLLC requirements are met. We then show on a possible radio setting for 5G, an example of radio resource dimensioning for different user traffic levels and we illustrate how the latency constraint may limit the allowable traffic for a given radio bandwidth.

Enabling URLLC warrants a major departure from average-based performance towards a clean-slate design centered on tail, risk and scale. We developed a framework for radio resource allocation for URLLC traffic in 5G use cases. We considered two classes of use cases, depending on the traffic generation profile. For deterministic packet generation, individual reservation of resources is needed for the first transmissions, while a pool of resources is reserved for retransmissions, scheduled by the base station. When traffic is sporadic, a contention-based scheme is adequate, where several replicas of each packet are randomly placed at different resources in order to increase the probability of success, despite possible collisions. In both cases, we derived analytical expressions for the reliability and used them to estimate the amount of resources needed for satisfying the reliability and latency targets. For use cases that allow the usage of both schemes, we showed how to choose between individual reservation and contention-based schemes for the lowest possible resource consumption. The approach takes into account various possible scenarios for the error probabilities on the basis of how many times the transmission of the same packet has taken place. It also considered the outcomes depending on whether there is enough space for all the packets that are lost to be retransmitted.

In the next generation of wireless communications the gNodeB establishes the link between the mobile network and the user equipment (UE). The resource blocks are required to be allocated in an optimal manner to improve the system efficiency without compromising on the reliability.

The benefits of artificial intelligence aided wireless systems equipped with machine learning were reviewed. The major families of machine learning algorithms and their applications in the context of next-generation networks were discussed. Existing machine learning algorithms can be categorized by the intended structure of the model. Most machine learning algorithms fall into the categories of supervised, unsupervised and reinforcement learning. In the first category, machine learning algorithms are provided with a labeled training data set. This set is used to build the system model representing the learned relation between the input, output and system parameters. In contrast to supervised learning, unsupervised learning algorithms are not provided



with labels (i.e., there is no output vector). Basically, the goal of an unsupervised learning algorithm is to classify the sample sets to different groups (i.e., clusters) by investigating the similarity between the input samples. The third category includes reinforcement learning algorithms, in which the agent learns by interacting with its environment (i.e., online learning). Finally, some machine learning algorithms do not naturally fit into this classification since they share characteristics of both supervised and unsupervised learning methods. These hybrid algorithms (often termed as semi-supervised learning) aim to inherit the strengths of these main categories, while minimizing their weaknesses. The primary intent of this work is to give insight into why machine learning is the future.

Linear regression model is used. Firstly, we cleanse the data collected and analyze data at the same time. . Then we choose user as the independent variable and the sale of throughput (response time) as the dependent variable to establish simple linear regression model for analysis. Programming language can also make data analysis in the field of data mining easier. Based on a real-time dataset of vehicular traffic data in an LTE network, the work presented here showed that using regression analysis in machine learning, the average resources consumed by various users can be examined and trained to predict throughput with an accuracy of more than 85%.

A dynamic approach would broaden the scope of the data which would account for various situations under which the system would function. Therefore to further improve upon accuracy of the predictions in the future, we would incorporate a multi parametric dynamic approach which would factor in several other parameters to ensure better prediction accuracy.