Evaluating Video Quality Degradation From Data Packet Losses in an LTE Environment

Alfonso Daniel Nombela Gordo

Luleå University of Technology
Master Thesis, Continuation Courses
Computer Science and Engineering
Department of Computer Science and Electrical Engineering
Division of Systems and Interaction
Evaluating Data Packet Loss Patterns and Resulting Video Quality Degradation in an LTE Network

M.Sc. Thesis Report

Alfonso Daniel Nombela Gordo

Luleå University of Technology
Dept. of Computer Science and Electrical Engineering
Div. of Systems and Interaction

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Abstract

Over a packet network such as Long Term Evolution (LTE), video communication will often be afflicted by packet losses. In fact, an important feature will be the actual packet loss pattern since two different patterns with a same loss rate could produce widely different degradation on a specific video. Moreover, one of the possible scenarios for a video communication is conversational video, for which a very important requirement is low latency. Therefore in this study, the definition of “packet loss” refers to the packets that arrive too late at application level.

Taking this definition into account, this master thesis evaluates the video quality degradation caused by the packet loss patterns in a LTE network for the specific scenario mentioned above. This evaluation is done by means of simulation using a LTE simulator and by identifying and modeling typical packet loss patterns from these simulations.

The simulations have shown that the patterns present a bursty distribution of the loss packets, i.e. if a packet is lost there is high probability that the following packets will be lost as well. Therefore, the patterns are modeled by a Logarithmic model, which is able to capture the burstiness of the distribution and which is shown to be more accurate than the well known Gilbert model.

The accuracy of the Logarithmic model is validated when the video quality is evaluated. Indeed, it is shown that the degradation caused by the simulation patterns has a high correlation with the degradation caused by the patterns modeled with the aforesaid method.
Preface

This work has been carried out between May and November 2009 at the company Ericsson AB in Luleå (Sweden).

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List of Acronyms

3GPP 3rd Generation Partnership Project
ACK Acknowledgement (in ARQ protocols)
AM Acknowledged Mode (RLC configuration)
ARQ Automatic Repeat-reQuest
BCCH Broadcast Control Channel
BCH Broadcast Channel
CCCH Common Control Channel
CQI Channel Quality Indicator
CRC Cyclic Redundancy Check
C-RNTI Cell Radio Network Temporary Identifier
DCCH Dedicated Control Channel
DL Downlink
DL-SCH Downlink Shared Channel
DRX Discontinuous Reception
DTCH Dedicated Traffic Channel
EDGE Enhanced Data rates for GSM Evolution and Enhanced Data rates for Global Evolution
eNB E-UTRAN NodeB
ePDG Evolved Packet Data Gateway
EPC Evolved Packet Core
E-UTRA Evolved UMTS Terrestrial Radio Access
E-UTRAN Evolved UMTS Terrestrial Radio Access Network
GSM Global System for Mobile communications
HARQ Hybrid ARQ
HSDPA High-Speed Downlink Packet Access
HSPA High-Speed Packet Access
HSUPA High-Speed Uplink Packet Access
IMT-2000 International Mobile Telecommunications-2000
IP Internet Protocol
ITU International Telecommunications Union
I-WLAN Interworking WLAN
KPI Key Performance Indicator
LTE Long Term Evolution
MAC Medium Access Control
MBMS Multimedia Broadcast Multicast Service
MCCH Multicast Control Channel
MCH Multicast Channel
MIMO Multiple Input Multiple Output
MME Mobility Management Entity
MOS Mean Opinion Score
MSE Mean Square Error
MTCH Multicast Traffic Channel
NACK Negative Acknowledgement (in ARQ protocols)
OFDM  Orthogonal Frequency Division Multiplexing
PAPR  Peak-to-Average Power Ratio
PBCH  Physical Broadcast Channel
PCCH  Paging Control Channel
PCH  Paging Channel
PDCCH  Physical Downlink Control Channel
PDCP  Packet Data Convergence Protocol
PDSCH  Physical Downlink Shared Channel
PDU  Protocol Data Unit
PEVQ  Perceptual Evaluation of Video Quality
PGW  Packet Data Network Gateway
PHICH  Physical HARQ Indicator Channel
PHY  Physical layer
PMCH  Physical Multicast Channel
PRACH  Physical Random Access Channel
PUCCH  Physical Uplink Control Channel
PUSCH  Physical Uplink Control Channel
QoS  Quality of Service
RACH  Random Access Channel
RAN  Radio Access Network
RAT  Radio Access Technology
RB  Resource Block
RLC  Radio Link Control
RRC  Radio Resource Control
RTP  Real Time Protocol
SAE  System Architecture Evolution
SAW  Stop-and-Wait
SC-FDMA  Single-Carrier Frequency Division Multiple Access
SDU  Service Data Unit
SGW  Serving Gateway
SI  System Information
SIMO  Single Input Multiple Output
SNR  Signal-to-Noise Ratio
SRB  Signaling Radio Bearer
TA  Tracking Area
TB  Transport Block
TCP  Transmission Control Protocol
TM  Transparent Mode (RLC configuration)
TTI  Transmission Time Interval
UE  User Equipment
UL  Uplink
UL-SCH  Uplink Shared Channel
UM  Unacknowledged Mode (RLC configuration)
UMTS  Universal Mobile Telecommunications System
VoIP  Voice-over-IP
WCDMA  Wideband Code Division Multiple Access
WLAN  Wireless Local Area Network
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1. Introduction

A growing amount of people spend much of their day traveling. Moreover, mobile communication technologies are becoming more and more used, either for business or leisure. Indeed, browsing Internet with the mobile phone has become a major activity and watching a Youtube video or a movie when traveling to work or to the school is increasingly applied.

Of course, all these services can be carried out through the current radio network system, the Universal Mobile Telecommunication System (UMTS). With its high data speeds (7.2 Mbit/s data rates from the outset, which may become 384 kbit/s in deployed networks), the user can consume mobile applications such as mobile TV and can have access to the World Wide Web. However, UMTS may present some problems, such as high power consumption and not very high spectral efficiency, and videos with a better quality (e.g. HD), which need higher data rates, are more and more demanded.

Due to all these reasons, 3rd Generation Partnership Project (3GPP) is specifying a new radio access technology, known as Long-Term Evolution (LTE). In the spring of 2005, the requirements and targets for LTE were set: it should provide downlink peak rates of at least 100 Mbit/s and 50 Mbit/s in the uplink and the main goals are improving spectral efficiency and services, making use of new spectrum and lowering costs. Due to these higher bit rates, the mobile TV service becomes even more attractive for LTE and will surely become more used.

As we can observe, LTE will introduce different advantages, e.g. low latency or high throughput. However, it will not be able to eliminate other phenomena that may lead to a distortion of the signal. Moreover, streaming video is especially sensitive to one of these phenomena: the packet losses. In this case, an important variable is the packet loss pattern since, for the same loss rate, two different loss patterns could produce widely different output images.

Therefore, this thesis aims to evaluate video quality degradation from data packet losses, where a packet is regarded as lost if arriving too late at application level, in typical LTE network scenarios by identifying typical packet loss patterns and creating a mathematical model to simulate these patterns.
2. Background

2.1. Problem Statement

The main objective of this thesis is to evaluate how real mobile network transmission problems degrade the quality of streaming video in a LTE network and to see if the more realistic packet loss models give a significant different result than the simpler models already used. Indeed, to be able to analyze how the quality of the video is degraded, typical loss patterns are identified and modeled.

2.2. Scope and Goals of the Thesis

Scope

Evaluating the video quality degradation in a real-life network demands a lot of work and in our case is almost “impossible”, since we have neither the necessary time nor sophisticated equipment to do such evaluation. Moreover, the LTE network will not be deployed until 2010/2011. So, to be able to carry out this work, we use a network simulator. First of all, we identify different scenarios that lead to packet losses in a radio network when streaming video from a certain base station to a certain cell phone, when using for instance a mobile video service. Changing and defining the settings and the parameters of the simulator, we are then able to run the simulations.

The next step in our project is based in a mathematical work. Based on the simulations, we have to create different mathematical models to be able to model the results obtained in the previous step. We also carry out some verification and calculations, for instance, the correlation between some statistics obtained from the data and some statistics predicted by the models. We are then able to compare them and evaluate media quality.

Finally, after this mathematical modeling part, we use a lightweight video impairment simulator called VideoBatchCoder (VBC) to get more visible results and to be able to see how the different packet loss patterns affect a real video clip. To do so, we parse the simulation patterns (i.e. the packet loss traces) to a packet loss file that fits the VBC, so we can decode video files with that
pattern and analyze them. Moreover, we also analyze how the patterns created from the different models would degrade the same video clip. Thus, a comparison between the degradation caused by the simulation patterns and the one caused from the models can be drawn.

**Goals**

The first goal is to obtain packet loss patterns for the different scenarios from the simulations carried out. The data that we can obtain from the simulator, which is in fact different files containing packet loss traces, is gathered and analyzed. The patterns are then identified.

The next goal is to characterize the packet loss pattern through the different mathematical models identified with high accuracy. The verification carried out should present high correlation between the models and the simulations.

The final goal is that the decoded video files from the VBC show that the video degradation caused by the packet loss patterns from the simulation is similar to the degradation caused by the models.

**2.3. Structure of the Thesis**

This thesis starts with the background to the actual market. The first chapter shortly introduces the reasons leading 3GPP to specify the new LTE system and the objective of the thesis. Then, Chapter 2 lays out the problem statement and the scope and goals of the thesis.

Next, in Chapter 3, the thesis gives an overview of the new LTE system. Firstly, the requirements specified by 3GPP for this new technology are introduced. Secondly, the main features and components of the LTE network are explained.

The focus then switches to analyzing the work carried out during this thesis. Chapter 4 therefore discusses the different LTE scenarios identified and analyzes the packet loss patterns obtained from the simulations conducted according to these scenarios. After that, Chapter 5 exposes the description of the different mathematical models identified and presents the verification and the comparison between them.

Chapter 6 finally lays out the evaluation of the video quality degradation thanks to the VBC and the comparison between the degradation caused by the simulation patterns and the modeled patterns.

Finally, Chapter 7 displays the different conclusions drawn from this thesis work and Chapter 8 sketches the hints for the work that can be conducted to continue this thesis.
3. Long Term Evolution (LTE): a technical overview

3.1. Introduction

Nowadays, thanks to the technology advancements the size and weight of the mobile terminals have been reduced dramatically and their functions have become more diversified. Indeed, apart from a phone, you can use that terminal as a camera or even as a small TV. Moreover, in parallel to the development of the mobile terminals, the number of different services delivered to the end users has also increased. In fact, there has been a recent increase of mobile data usage and emergence of new applications such as mobile TV, Web 2.0 or Multimedia Online Gaming. Thus, the mobile communications technologies need to develop as well to be able to stay competitive and to meet the new demands.

All these reasons have motivated the 3rd Generation Partnership Project (3GPP) to specify a new radio access technology: Long Term Evolution (LTE). LTE is the latest standard in the mobile network technology tree that previously realized the GSM/EDGE and UMTS/HSxPA network technologies (Figure 3.1). While initial deployments of LTE are expected by 2010, commercial use will not be available until one or two years later.

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**Figure 3.1**: 3GPP standards evolution.
LTE is clearly influenced by the WCDMA and the HSPA work in 3GPP and is expected to substantially improve end-user throughputs, current capacities, achieve low delay and provide full mobility for an improved user experience. Furthermore, the success of the Internet and the IP-based services delivered over the Internet requires the mobile communications systems to offer more and more IP-based services. So, LTE should be able to provide support for IP-based traffic and voice traffic will be supported mainly as Voice over IP (VoIP).

At the same time as LTE, 3GPP is specifying an evolved core network, the Evolved Packet Core (EPC). This new network architecture will be able to support the Evolved UMTS Terrestrial Radio Access Network (E-UTRAN), LTE’s radio access, through a reduction in the number of elements and with a simpler functionality. The work done with the EPC is called System Architecture Evolution (SAE) study.

To be able to have a successful deployment with this new network system, LTE has set clear performance requirements that rely on physical layer technologies, such as Orthogonal Frequency Division Multiplexing (OFDM) and Multiple Input Multiple Output (MIMO) systems. The main goals are to achieve spectrum flexibility, minimize the system and user complexities and to be able to co-exist with other already existing 3GPP Radio Access Technologies (RATs) such as GSM and WCDMA/HSPA.

3.2. Design targets

As we have already explained, 3GPP set very aggressive requirements for the development of both LTE and SAE. In this part, the different targets are analyzed and discussed.

3.2.1. LTE design targets

The requirements and targets for LTE were split into different areas, as is documented in [1]. We are going to describe each of these areas.

3.2.1.1. Capabilities

The first requirements for LTE were set in terms of capability. Indeed, as seen before, the system is expected to support different types of services, such as VoIP, video streaming or web browsing. Consequently, LTE had to be designed to have a high peak data rate. So, the targets for downlink and uplink peak data rate requirements were set to 100 Mbit/s and 50 Mbit/s respectively when operating in 20 MHz spectrum allocation, the expected bandwidth capability of a User Equipment (UE).

Furthermore, to be able to support the different types of services we have described, another important requirement that has to be met by the system is low latency. In fact, the latency requirements are divided into control plane requirements and user plane requirements. The control plane latency is defined as the transition time to pass from a non-active terminal state to
an active state where the mobile terminal can send or receive data. In this case, the requirement for the transition time is that it has to be smaller than 100 ms. Besides that, 3GPP defined another control plane requirement: LTE should support at least 200 users with their mobile terminal in the active state, when operating in 5 MHz spectrum allocation.

On the other hand, the user plane latency is defined as the time it takes to transmit an IP packet from the terminal to the RAN edge node or vice versa. Therefore, the user plane latency requirement is that this time should be smaller than 5 ms.

3.2.1.2. System performance

A spectrum efficiency target has also been specified. It is defined as the system throughput per cell (measured in bits/MHz/cell) and in the LTE Downlink will be 3 or 4 times of that of Release 6 HSDPA, whereas in the Uplink it will be 2 or 3 times of that of Release 6 HSUPA.

Mobility requirements have also been defined. A different performance is targeted depending on the speed of the mobile terminal: at low speeds (0-15 km/h), maximal and optimal performance should be achieved; for speeds up to 120 km/h, high performance is targeted and for speeds above 120 km/h, the system should be able to maintain the connection. The mobility support should permit a maximum speed of 350 km/h (or even 500 km/h depending on frequency band).

Another aspect that has been specified is the coverage requirement, which refers to the cell sizes (radius). It was set that, for coverage of 5 to 100 km the other requirements defined (mobility, spectrum efficiency, etc.) should be met, tolerating a slight degradation for cells with a radius bigger than 30 km.

3.2.1.3. Spectrum flexibility

Since the global spectrum situation becomes more complex each day, a requirement on spectrum flexibility needs to be specified. Indeed, LTE needs to be able to operate in existing IMT-2000 frequency bands, which implies co-existence with the systems that are already working in those bands (GSM, WCDMA/HSPA …).

So, the spectrum flexibility requirement defines the need for LTE to be scalable in the frequency domain and targets the system to operate in 1.25, 1.6, 2.5, 5, 10, 15 and 20 MHz spectrum allocations. Moreover, LTE should be able to operate in unpaired as well as paired spectrum.

3.2.2. SAE design targets

As for LTE, the SAE requirements were divided into different areas [2]. We are going to analyze the most important ones.

As we have already explained, LTE will need to co-exist with other already existing 3GPP radio access technologies. Therefore, the SAE system should be able to operate with other types of
radio access network, i.e. not only with LTE. Moreover, a mobile terminal should be able to move between the different radio access systems so mobility functions should be present in the SAE.

Clearly, the SAE requirements define that the traditional services such as voice, video, messaging and data file exchange will be supported. Moreover, based on the requirement of IP connectivity support, any service based on IP will be supported.

Moreover, another important requirement for SAE is roaming, including inbound and outbound roaming to other SAE networks.

Apart from that, SAE system should be packet based, supporting real-time and conversational class traffic and the architecture should minimize the presence of “single points of failure”.

3.3. LTE radio access

After analyzing the different targets set for LTE and SAE, we are going to present in this part the most important aspects and components of the LTE system.

3.3.1. Transmission scheme: OFDM and SC-FDMA

The Evolved UMTS Terrestrial Radio Access (E-UTRA) interface makes exclusive use of shared channels for both data and signaling transfer. In this respect, E-UTRA is similar to the 3GPP HSPA; however, the selected radio access technology is very different to HSPA. While HSPA uses WCDMA, the E-UTRA uses Orthogonal Frequency Division Multiplexing (OFDM).

OFDM splits the available system bandwidth into hundreds of narrowband “sub-carriers” [9]. This means that a high bit rate data stream to a certain UE is split by the evolved NodeB (eNB) into a large number of narrowband, low bit rate data streams. The received parallel sub-carriers are then multiplexed by the UE in order to re-create the original high data bit rate data stream.

The OFDM technology has several advantages:

- It has better spectral efficiency, i.e. more information can be sent using the same system bandwidth as compared to a single-carrier system.
- OFDM provides access to the frequency domain, thereby enabling an additional degree of freedom to the channel-dependent scheduler compared to HSPA.
- OFDM supports flexible spectrum allocations. Indeed, the system bandwidth can be expanded in increments, by “adding” more sub-carriers, as more spectrum becomes available to the operator.
- It has a better performance under multipath fading conditions. Multipath effects lead to frequency-selective fading, which is more damaging to a wideband signal
than to a narrowband signal. So, OFDM presents more robustness to this frequency-selective fading.

However, there are drawbacks with OFDM as well. One important drawback is that an OFDM signal has a very high peak-to-average power ratio (PAPR), which leads to very inefficient use of power amplifiers, and hence high power consumption, in a mobile terminal. Therefore, OFDM is only used for the LTE downlink transmission scheme.

On the other hand, the LTE uplink is based on a variant of OFDM that reduces PAPR, which is called Single-Carrier Frequency Division Multiple Access (SC-FDMA). The use of this single-carrier transmission therefore allows for more efficient usage of the power amplifier (since the smaller the PAPR, the higher the average transmission power can be for a given power amplifier), which translates into an increased coverage.

3.3.2. Multiple antenna support

LTE supports the use of multiple antennas at both the base station and the UE as an integral part of the E-UTRA standard. The two main reasons to use multiple antennas are:

- To transmit more information over the radio interface without using more bandwidth than a single antenna system; the number of antennas used increases the system capacity in a linear manner.

- To transmit the same information, with the same bit rate as a single antenna system, but with less output power (higher reliability).

Furthermore, there are different ways to use multiple antennas, each one having a different purpose:

- Multiple receive antennas can be used for receive diversity. This diversity can be used against fading, but also to suppress interference. In this case, where the transmitter has one antenna, the system is called SIMO (Single Input Multiple Output). SIMO system, illustrated in Figure 3.2, is widely used.

![Figure 3.2: SIMO system.](image)
• Multiple transmit antennas at the base station can be used for transmit diversity and different types of beam-forming, which will improve the received SNR, system capacity and coverage.

• A third way to make use of the multiple antennas technique is by combining the two previous ones, i.e. using multiple antennas at both the transmitter and the receiver, known as MIMO (Multiple Input Multiple Output) or spatial multiplexing. MIMO is also supported by LTE and leads to an increased data rate in bandwidth-limited scenarios. MIMO system is illustrated in Figure 3.3.

![MIMO system](image)

**Figure 3.3:** MIMO system.

### 3.3.3. LTE radio interface architecture

Similar to most other modern communication systems, the LTE radio interface architecture is structured into different protocol layers. Although the protocols are by name and function similar to the protocol architecture used for WCDMA/HSPA, there are two main differences:

• Firstly, the LTE protocols are, on the network side, all located in the eNB (which is not the case for WCDMA/HSPA protocols).

• Secondly, the LTE protocols have been all simplified in terms of complexity and functionality, when compared to the WCDMA/HSPA protocols.

Furthermore, to be able to offer services between the different protocols, there is also a channel architecture. The LTE protocol and channel architecture can be seen in Figure 3.4 and will be discussed in this part.
3.3.3.1. Radio Resource Control (RRC)

The RRC protocol [3] is responsible for all control signaling exchange between the UE and the eNB. The functionality of this protocol is very similar to the WCDMA/HSPA RRC protocol. We will explain briefly some RRC procedures:

- **Broadcast of System Information (SI)**, whose purpose is to provide the UEs with essential parameters needed for communication with the network.

- **RRC Connection Establishment**, which moves the UE from RRC Idle state to the RRC Connected state (see LTE states below). This RRC connection is always needed whenever the UE wants to send/receive control signaling or data to/from the network.

- **Management of Radio Bearers**, used for establishment, reconfiguration and release of user plane radio bearers, including configuration of ARQ (RLC) and HARQ (MAC).

- **Handover Control**, which includes the control signaling to execute hard handovers between eNBs or between an eNB and some other RAT.
3.3.3.2. Packet Data Convergence Protocol (PDCP)

The PDCP [4] is responsible for IP header compression to reduce the number of bits necessary to transmit over the radio interface, access stratum security and for in-sequence delivery of user plane packets during handover.

Moreover, the protocol is also responsible for ciphering and integrity protection of the transmitted data, and at the receiver side it performs the corresponding deciphering and compression operations. There is one PDCP entity per radio bearer configured for a mobile terminal.

3.3.3.3. Radio Link Control (RLC)

The LTE RLC protocol [5] offers link control over the radio interface for user data and control signaling and is responsible for segmentation of IP packets (known as RLC SDUs) from the PDCP into smaller units (known as RLC PDUs). It can also handle retransmission of erroneously received PDUs, as well as duplicate removal and concatenation of received PDUs. Indeed, one UE may use multiple RLC entities simultaneously, with each entity configured to operate in one of three modes: Acknowledged Mode (AM), Unacknowledged Mode (UM) or Transparent Mode (TM).

When working in AM, the RLC layer guarantees that all PDUs delivered to the higher layer are received without errors. To achieve this, the RLC protocol uses RLC-level acknowledgements (ACK) and retransmission requests (NACK). In fact, a retransmission protocol operates between the RLC entities in the receiver and the transmitter and by monitoring the incoming sequence numbers, the receiving RLC can identify missing PDUs. ACK or NACK are indicated with status reports (RLC Control PDUs) sent from receiver to transmitter, which will take the appropriate action. AM is typically used for TCP-based services where having error-free data is very determinant.

On the other hand, in UM there are no acknowledgements or retransmissions – hence, delivery of error-free data cannot be guaranteed. Any erroneous RLC PDUs received are discarded or delivered to higher layers with an indication that they contain errors. UM is typically used for services where delivery of data without errors is of less importance compared to short delivery time (such as VoIP). Finally, TM is used for minimum protocol overhead and only for specific purposes such as random access.

As can be seen in Figure 3.4, the RLC offers services to the user plane (PDCP protocol) in the form of Radio Bearers (RB). The RLC mode to use is configured with the Radio Bearer Establishment procedure.

The service the RLC provides to the control plane (RRC protocol) is called a Signaling Radio Bearer (SRB), which is configured during the RRC Connection Establishment procedure.
3.3.3.4. Medium Access Control (MAC)

The main functions of the MAC protocol [6] are to handle multiplexing of one or several upper layer PDUs onto transport blocks, to perform uplink and downlink resource allocation through dynamic scheduling and to perform error correction through hybrid ARQ (HARQ) retransmissions. The scheduling functionality is located in the eNB, which has one MAC entity per cell, for both uplink and downlink and the hybrid-ARQ protocol part is present in both the transmitting and receiving end of the MAC protocol.

Logical channels and Transport channels

As we can observe in Figure 3.4, the MAC layer offers services to the RLC in the form of Logical channels and it uses services from the physical layer in the form of Transport channels. We are going to describe and explain the differences between these two types of channels.

Firstly, a logical channel describes the type of information to be transmitted and is an information stream dedicated to the transfer of a specific type of information over the radio interface. There is a general classification of logical channels into two groups: Control channels (used for transmission of control and configuration information necessary for operating an LTE system) and Traffic channels (used for the user data). The different logical channels are:

- **Broadcast Control Channel (BCCH):** used for broadcasting system control information from the network to all mobile terminals in a cell (downlink channel).
- **Common Control Channel (CCCH):** bi-directional channel used for transmitting initial RRC control signaling between the UE and eNB.
- **Dedicated Control Channel (DCCH):** point-to-point bi-directional channel for sending dedicated RRC control signaling between the UE and the eNB. It is used for individual configuration of mobile terminals.
- **Multicast Control Channel (MCCH):** point-to-multipoint downlink channel used for transmitting Multimedia Broadcast Multicast Service (MBMS) control information from the network to the UE (only used by UEs that receive MBMS transmissions).
- **Paging Control Channel (PCCH):** downlink channel that carries paging information and system information change notifications (used for paging when the network does not know the location cell of the UE).
- **Dedicated Traffic Channel (DTCH):** point-to-point channel dedicated to one UE (uplink or downlink or both) for transmission of user data.
- **Multicast Traffic Channel (MTCH):** point-to-multipoint downlink channel for transmission of multimedia traffic (e.g. mobile TV) from the network to the UE (only used by UEs that receive MBMS transmissions).
Secondly, the transport channels describe in what format the information is to be transmitted. Therefore, different transport channels are defined by how and with what characteristics the information is transmitted on the physical layer.

Information on transport channels is delivered to/from the physical layer in the form of Transport Blocks (TB). One or two TB are delivered per Transmission Time Interval (TTI), whose length for LTE is 1 ms for most transport channels. With each TB, a Transport Format (TF) is associated; it specifies how the T is to be transmitted over the radio interface and is a combination of TB size (in bits), TTI length and layer 1 channel coding and modulation selected for a given transmission.

Then, the different transport channels are:

- **Broadcast Channel (BCH):** it has a fixed, pre-defined transport format and carries part of the SI in a cell. It is used for transmission of the information on the BCCH logical channel.

- **Multicast Channel (MCH):** carries MBMS data and control information in case of an MBMS-dedicated cell. It is characterized by a semi-static transport format and semi-static scheduling.

- **Paging Channel (PCH):** used for transmission of paging information on the PCCH logical channel. It supports UE discontinuous reception (DRX) to enable UE power saving.

- **Downlink Shared Channel (DL-SCH):** it is the main downlink resource in E-UTRA. It carries data (DTCH or MTCH) and signaling (BCCH, CCCH, DCCH or MCCH). The DL-SCH uses HARQ, channel dependent packet scheduling and adaptive modulation and coding.

- **Random Access Channel (RACH):** uplink channel used to carry control information from the UE to the eNB. It is used for initial access, when the UE is not known in the eNB.

- **Uplink Shared Channel (UL-SCH):** it is the main uplink resource in E-UTRA. It carries data (DTCH) and signaling (CCCH and DCCH). The UL-SCH uses HARQ, channel dependent packet scheduling and adaptive modulation and coding.

Furthermore, the MAC layer performs multiplexing of different logical channels and mapping of the logical channels to the appropriate transport channels. The mapping between the logical channels and the transport channels is shown in Figure 3.5 (both downlink and uplink).
Dynamic Scheduling

Since time-frequency resources are dynamically shared between users in both uplink and downlink in LTE, the scheduling function manages DL-SCH and UL-SCH radio resources between UEs. The scheduler is part of the MAC layer and controls the assignment of uplink and downlink resources. To have a correct performance, the scheduling algorithm should take into account different aspects such as availability of radio resources, data stream priority levels, amount of data awaiting transmission or how long time since a certain user was last served.

The scheduling functionality is different for LTE downlink and uplink. Indeed, for the downlink scheduler, the basic principle is to determine which UE can receive data from the DL-SCH and on what resources each 1 ms TTI. Multiple terminals can be scheduled in parallel, in which case there is one DL-SCH per scheduled terminal, each dynamically mapped to a set of frequency resources.

For the uplink scheduler, the function is similar to the downlink, but in this case it determines which UE is able to transmit data on the UL-SCH and on which uplink resources. The shared resource controlled by the eNB scheduler is time-frequency resource units.
Hybrid ARQ

LTE hybrid ARQ is used to provide robustness against transmission errors and to enhance capacity of the system. The HARQ protocol is part of the MAC layer and one HARQ entity within MAC handles the HARQ functionality for one user.

The LTE HARQ protocol is of the “Stop-and-Wait” type (SAW), i.e. it is not allowed to transmit a PDU with sequence number n until the PDU with sequence number n-1 is acknowledged. When a transport block is received, the receiver tries to decode it and informs the transmitter whether the decoding was successful or if a retransmission is required through a single ACK/NACK.

With HARQ, the impact on end-user performance from erroneously received packets is minimized since retransmissions can be rapidly requested after each packet transmission.

3.3.3.5. LTE Physical Layer (PHY)

The physical layer is responsible for protecting data against channel errors using adaptive modulation and coding schemes based on channel conditions. It is also responsible for mapping of the signal to the appropriate time-frequency resources (by maintaining frequency and time synchronization) as well as for performing RF processing, including modulation and demodulation (QPSK, 16-QAM and 64-QAM are the DL and UL schemes), and multi-antenna processing.

The physical layer encodes the transport channel data to physical channels, which are defined by [8] as “a set of resource elements carrying information originating from higher layers”. The physical channels are divided into downlink physical channels and uplink physical channels:

- Physical Downlink Shared Channel (PDSCH): it is the main downlink radio resource in a cell. It carries the DL-SCH and PCH TB and uses QPSK, 16-QAM or 64-QAM modulation. Since it is a shared resource, all downlink transmissions must be explicitly addressed to the receiving UE, what is done on the PDCCH (see below).

- Physical Downlink Control Channel (PDCCH): it informs the UE about the resource allocation of PCH and DL-SCH, and HARQ information related to DL-SCH. It is also used for allocation of uplink resources on PUSCH/PUCCH. It uses QPSK modulation.

- Physical Broadcast Channel (PBCH): this downlink channel carries the BCH transport block and uses a TTI of 40 ms. It uses QPSK modulation.

- Physical HARQ Indicator Channel (PHICH): the purpose of this downlink channel is to transmit Hybrid ARQ ACK/NACKs related to uplink transmissions on the PUSCH. It uses QPSK modulation.

- Physical Multicast Channel (PMCH): it carries the TB from the MCH. It uses QPSK, 16-QAM or 64-QAM modulation.
• **Physical Uplink Shared Channel (PUSCH):** it is the main uplink radio resource. It carries the UL-SCH TB and uses QPSK, 16-QAM or 64-QAM modulation. Since it is a shared resource, all uplink transmissions must be explicitly allocated to a given UE, what is done on the PDCCH.

• **Physical Uplink Control Channel (PUCCH):** it carries Hybrid ARQ ACK/NACKs in response to downlink transmissions, uplink scheduling requests and Channel Quality Indicator (CQI) reports. It uses BPSK or QPSK modulation.

• **Physical Random Access Channel (PRACH):** this uplink channel carries the random access preamble, which is used for the random access procedure when the UE wishes to initiate uplink transmissions and does not have a valid uplink grant.

As we have already explained, the physical layer is responsible for performing different types of processing on the data delivered by the transport channels. In Figure 3.6, a simplified overview of the physical layer processing for DL-SCH and UL-SCH is plotted.

![Figure 3.6: Physical Layer processing for a) DL-SCH and b) UL-SCH.](image)

We can observe that the physical layer processing is similar for the DL-SCH and the UL-SCH. In fact for both channels, before the scrambling part, a 24-bit Cyclic Redundancy Check field (CRC) for error detection is attached to the transport block received and each such combined CRC-TB is separately coded.

However, we can also appreciate some differences. For instance, for downlink transmission an OFDM signal is generated while in the UL-SCH a SC-FDMA signal is generated, according to the different technologies used.
3.3.3.6. LTE states

In the LTE system, there are three different states for a certain UE: the LTE_Detached state, the LTE_Active state and the LTE_Idle state.

![LTE states diagram](image)

**Figure 3.7:** LTE states. Adapted from [9].

As can be seen in Figure 3.7, a certain mobile terminal enters the LTE_Detached state at power up. In this state, there is no information known about the UE and it is not known to the network. Furthermore, in this state no data or signaling transfer is possible. So to be able to get any kind of transmission, the terminal needs to enter the LTE_Active state, what is done by registering with the network.

In the LTE_Active state, the UE is then registered and the network knows the cell to which the UE belongs, allowing data or signaling to be sent or received. In this state, the UE is always allocated with a cell specific identifier, the Cell Radio Network Temporary Identifier (C-RNTI), used for signaling purposes between the mobile terminal and the network.

In case of inactivity, the UE will enter the LTE_Idle state. In this state, the mobile terminal is in power-conservation and typically it is not transmitting or receiving data. In the LTE_Idle state, the UE does not inform the network about each cell change. However, the network knows the location of the UE to the granularity of a few cells, called the Tracking Area (TA) and when there is a UE-terminated call, the UE is paged in its last reported TA.
3.4. System Architecture Evolution (SAE)

As mentioned before, 3GPP is also working on a System Architecture Evolution (SAE) to be able to support the new packet-data capabilities provided by the LTE radio interfaces [10]. The result of this work is an evolved core network, called the Evolved Packet Core (EPC), with different functionalities and elements from the already existing core network. The EPC network architecture is outlined in Figure 3.8.

![Figure 3.8: EPC network architecture.](image)

This new architecture is specified to optimize network performance, improve cost efficiency and facilitate the uptake of emergent new IP-based services. In this part, we will explain briefly the main functional elements of the network architecture.

**Mobility Management Entity (MME)**

The MME is the key control-node for the LTE access network. It manages and stores contexts relating to UEs in both Idle and Active state and handles mobility management procedures such as paging, registration/de-registration, Tracking Area updates and security procedures (authentication, allocation of temporary identities …). Indeed, it is responsible for choosing the SGW for a UE at the initial attach and for authenticating the user by interacting with the HSS.

The MME also provides the control plane function for mobility between LTE and other radio access systems such as 2G or 3G systems.
**Serving Gateway (SGW)**

The SGW terminates the downlink data path for UEs in the Idle state and initiates paging to the MME when downlink data arrives for the UE. It also manages and stores UE contexts, such as user IP-address or network internal routing information.

During and after a handover to UTRAN, the SGW acts as the mobility user plane anchor, routing and forwarding user data packets to the SGSN. It also acts as the anchor for mobility between LTE and other 3GPP technologies.

**Packet Data Network Gateway (PGW)**

The PGW is the connection between the EPC and external packet data networks, and provides this connectivity by being the point of entry and exit of traffic for the UE. It is also responsible for allocation of user IP-addresses and for packet filtering for each user.

The PGW also acts as the user plane anchor for mobility between E-UTRAN and other IP-access networks. These other IP-access networks are divided into 3GPP IP-access (which is the Interworking WLAN, I-WLAN), trusted non-3GPP IP-access and non-trusted non-3GPP IP-access. In this ‘other IP-access networks’ group we can find, for instance, xDSL, CDMA2000 or WiMAX.

**Evolved Packet Data Gateway (ePDG)**

The ePDG performs access authentication when a mobile terminal tries to connect to the home domain. If needed, it performs QoS authorization and generates charging information for the packet data session.
4. Packet Loss Patterns

Video communications over a packet network such as LTE will be of course very sensitive to the packet losses that may occur during a certain transmission of data, and the distortion that may experience the video will be dependent on the different characteristics of the losses. In fact, a key feature will be the packet loss pattern and the resulting distortion will be mostly determined by the properties of these patterns [11]. For instance, a more “bursty” distribution, where the lost packets arrive after each other, is often perceived as better than an even one.

To be able to identify different kinds of loss pattern, the transmission of data should experience some packet losses, i.e. this exchange of data should take place in a certain scenario that leads to packet losses in the LTE network.

In this section, we firstly outline the different LTE scenarios identified. After that, we discuss the parameters of the different simulations we run. Finally, we explain the output information we get from the simulations, that is, we analyze the different packet loss patterns we are able to identify.

4.1. Network Scenarios

As we have already explained, 3GPP specified the LTE system with several different properties. Indeed, it defined capacity requirements as well as performance requirements. Because of that, a certain LTE communication between a server and a mobile TV user can take place in very different situations with remarkably different results. To identify network scenarios where we could get packet losses, we analyzed the specified features of the new radio access technology.

A first aspect that can lead to varied scenarios is the number of users. As stated above, the LTE system should support at least 200 users per cell. So, depending on the number of users in the cell, the load of the system and the end-user throughput will vary, carrying into a different number of packet losses. Thus, we are able to vary the number of mobile TV users in a cell downloading a certain video to observe how the system behaves in different situations.

Another important characteristic of the LTE system that can be taken into account to identify different scenarios is the capacity of using multiple antennas at both the base station and the
terminal. The two main combinations within this multiple antenna technology are by using one transmit antenna and multiple receive antennas (SIMO) and by using multiple transmit antennas and multiple receive antennas (MIMO). Therefore, depending on what kind of combination you use, the properties and the performance of the system can change.

In addition to that, another feature that may lead into a different number of packet losses and that was also part of the requirements in the 3GPP specifications is the cell sizes. Indeed, the throughput may be better for a user that is close to the base station than for a user that is farther; moreover, the bigger the coverage of the cells the farther a user can be. So, in our case, by varying between the standards 3GPP case 1 and 3GPP case 3 [12], which implies basically a change in the cell radius, we can obtain different scenarios.

In consequence, by combining these variable aspects, we are able to create LTE scenarios in which the system has different characteristics and properties and where we are able to identify a changing number of packet losses and varied loss patterns. For instance, a first simulated scenario is a system based on the 3GPP case 1 standard, using MIMO technology and having 100 users making use of the DL; another one is a system based on 3GPP case 3, using SIMO technology and having 150 users in the DL; etc.

As we explain below, the objective when identifying different packet loss patterns is to gather these patterns for different loss levels (1, 5 and 10%) and to analyze their characteristics. Therefore, by changing e.g. the number of users in a cell, the load level in the system is adjusted in order to reach these target loss rates.

The variation of these three parameters, combined with some fixed parameters that are explained in the next section, create the network scenarios used during the simulations.

4.2. Simulation Settings

As explained before, in order to carry out this study a LTE simulator was used. We will then explain the different settings and parameters applied to the simulator.

4.2.1. General Simulator Settings

When running the simulations, several fixed parameters were settled. Some of the main system settings are listed below:

- 7 base stations, with 3 cells per base station (21 cells in total)
- 3 km/h UE speed (with straight line movement)
- Total bandwidth of 5 MHz for the frequency band
- Round Robin scheduling algorithm

With regard to the cell settings, they change depending on the scenario in which we are working:
- Cell radius of 166.666 m when simulation based on 3GPP case 1
- Cell radius of 577.3333 m when simulation based on 3GPP case 3

Likewise, some eNB specifications will be different for a scenario where the system uses MIMO technology (each eNB has two transmit antennas) than for a scenario where the system uses SIMO technology (each eNB has just one transmit antenna). In both cases, each UE has two receive antennas. We can therefore observe that the exchange between MIMO and SIMO is just applied for the downlink. For the uplink, we use always SIMO (one transmit antenna at the UE and two receive antennas at the eNB).

Besides, the protocol used to transfer the packets is the UDP protocol and the RLC mode used in this case is RLC AM.

As regards the HARQ settings, the maximum time allowed for retransmissions is 60 ms and each retransmission takes more or less 8 ms. Therefore, the maximum number of HARQ retransmissions will be approximately 7.

4.2.2. User Settings

In this study, only one type of user is present in a certain scenario, which is the mobile TV user. Moreover, two variants of this user are implemented during the simulations: a downlink user, which is having a downlink transmission and thus is making use of the DL channel, and an uplink user, which is making use of the UL channel. Even if only the downlink performance is of interest here, we decided to include an uplink user to make the simulation more realistic and to include other kind of load to the system.

During the first simulations we ran, what was modeled was a streaming video user, i.e. a usage scenario where each user requests to receive a specific video clip (sending the necessary request to the server), then a short buffering period follows, after which the play-out of the video begins. This kind of user has also re-buffering functionality, which means that if a certain frame is missing during the play-out of the clip (i.e. that the buffer is empty) the user stops receiving frames and playing out the video and waits for the re-buffering (i.e. waits for the buffer to have enough frames to play again). Therefore, we realized after some simulations that, due to the RLC AM mode, it was impossible to achieve any loss – hence impossible to identify any packet loss pattern.

Consequently, another kind of mobile TV user was modeled in the simulations, the real-time video user. In this case, what is modeled is not on-demand TV, but rather a usage scenario where the user does not request any traffic. Instead, the server initiates the conversation and the user plays out frames as soon as it receives them. For this kind of user, a certain packet will be lost if the delay of this packet is greater than the maximum allowed delay defined in the simulations. Therefore, the term “packet loss” henceforth refers to a packet that arrives too late at application level.

The most relevant settings are shown in Table 4.1.
Table 4.1: Mobile TV users settings.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Lifetime</td>
<td>30 s</td>
</tr>
<tr>
<td>Maximum allowed delay</td>
<td>150 ms</td>
</tr>
<tr>
<td>Number of users during a simulation</td>
<td>Fixed</td>
</tr>
<tr>
<td>Number of users in the UL</td>
<td>50</td>
</tr>
<tr>
<td>Number of users in the DL</td>
<td>70 - 180</td>
</tr>
</tbody>
</table>

As we can see in Table 4.1, the number of users during a certain simulation is maintained, i.e. the number of initial users does not change during the whole time of the simulation. Moreover, when each user is created (in this case, only at the beginning) they are given a random location in the system and thereafter move in a random direction with a constant speed of 3 km/h.

In addition, the number of users making use of the DL channel is different from simulation to simulation, depending on which scenario we want to simulate. At the same time, the number of users in the UL will be the same for all the simulations.

4.2.3. Video Settings

In order to simulate the video received by the mobile TV users, video frame trace files are used. They are generated from real video clips and they contain the sizes of all video frames from the original clip. Therefore, to create a realistic simulation of the video stream, frames of these sizes are transmitted at the same frame rate as the original clip.

In this study, the simulations were run using a trace file containing frame traces from an MTV music video. This video was encoded with the x264 encoder with the H.264 baseline profile and 25 frames per group of pictures. In Table 4.2, the main settings of the video trace file are summarized.

Table 4.2: MTV video trace file settings.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mean bit rate</td>
<td>403 kbps</td>
</tr>
<tr>
<td>Codec</td>
<td>H.264</td>
</tr>
<tr>
<td>Frame rate</td>
<td>25 frames per second (fps)</td>
</tr>
<tr>
<td>Resolution</td>
<td>CIF (352 x 288)</td>
</tr>
</tbody>
</table>
The traffic pattern is illustrated in Figure 4.1:

In each simulation, all users will be streaming the same video clip and, according to the lifetime of each user, the video duration is set to 30 s. Furthermore, in order to avoid synchronization of all the video streams, each user begins receiving packets at a random starting point within the traffic trace and, to avoid the synchronized transmission of frames for all users, a small random offset is used before the first frame is transmitted (offset range is 0 to 40 ms).

4.3. Simulation Results

4.3.1. Procedure

After applying the correct settings and parameters described in the previous section, we conducted a series of simulations in which we varied the three parameters already explained: MIMO/SIMO, number of users in the DL and 3GPP case 1/case 3. Moreover, the simulator enables you to identify the packets that are lost for every user present in the system. So, in order to obtain a more realistic result, we did not analyze the behavior of the same user in each simulation, but instead ran several simulations for several users.
We knew beforehand that in packet transmissions over many real networks, the packet losses occur in bursts separated by loss-free gap [13] [14]. Therefore, expecting a similar behavior for the LTE network, we analyzed the loss patterns focusing on the loss run lengths, i.e. the length of the sequences of consecutively lost packets. In fact, a certain burst loss begins with a packet that has been lost, when the previous packet has been successfully received, and ends with a lost packet when the following packet is successfully received; the length of the burst is the number of consecutive lost packets. Therefore, a single lost packet is a burst loss of length one.

The objective in this study was to analyze the properties of the packet loss patterns for different loss rates. Thus, we looked for the patterns obtained from the simulations for several target loss rates: 1, 5 and 10% losses. Besides, for each loss rate we gathered the pattern given by a system using MIMO and the pattern given by a system using SIMO to evaluate if these two kinds of implementations induced remarkable differences in the loss patterns.

4.3.2. Results

As a result, we studied the properties of the different patterns by calculating the loss run lengths for each one of them. In Tables 4.3, 4.4 and 4.5, we summarize the contribution of burst losses of various lengths for the desired loss rates and for MIMO and SIMO. Furthermore, to illustrate our simulations results and to have a brief overview of the packet loss distribution, we include images of the packet loss pattern for the different loss rates and for MIMO and SIMO, where a darker mark indicates a lost packet. Since each image is taken from one simulation, they do not represent an average result and should be used just as indication.

Table 4.3: Contribution of burst losses (in percents) of length n for 1% packet loss.

<table>
<thead>
<tr>
<th>n (packets)</th>
<th>1</th>
<th>2</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>1% loss rate</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>MIMO</td>
<td>93.33</td>
<td>6.67</td>
</tr>
<tr>
<td>SIMO</td>
<td>94.12</td>
<td>5.88</td>
</tr>
</tbody>
</table>
For a packet loss rate of 1%, the pattern does not seem to be bursty. Indeed, we can observe from Table 4.3 that only burst losses of length 1 and 2 packets are present in the pattern, being burst losses of length 1 much more important than burst losses of length 2. In addition, by looking on Figure 4.2, it seems that we only have single lost packets and that the losses occur at random instants.

Apart from that, we can also appreciate from Table 4.3 and Figure 4.2 that the pattern obtained from the simulation using MIMO is very similar to the pattern obtained from the simulation using SIMO.

Table 4.4: Contribution of burst losses (in percents) of length n for 5% packet loss.

<table>
<thead>
<tr>
<th>n (packets)</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
</tr>
</thead>
<tbody>
<tr>
<td>5% loss rate</td>
<td>MIMO</td>
<td>83.12</td>
<td>7.79</td>
<td>3.9</td>
<td>2.6</td>
<td>1.3</td>
<td>1.3</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td></td>
<td>SIMO</td>
<td>73.68</td>
<td>17.11</td>
<td>5.26</td>
<td>0</td>
<td>1.32</td>
<td>0</td>
<td>1.32</td>
<td>0</td>
</tr>
</tbody>
</table>
However, when the loss rate increases, the burstiness of the loss pattern becomes appreciable. In Table 4.4, it can be observed that the contribution of burst losses of length 1 decreases in favor of burst losses of bigger lengths. From Figure 4.3 we can perceive that the losses do not seem to take place at random instants any more, but the lost packets tend to arrive after each other.

With regard to the difference between MIMO and SIMO, it seems that the pattern given by the scenario using SIMO is slightly more bursty, since we can encounter burst losses of length 7 or 9 packets. We can also observe from Figure 5.2 that the loss bursts occur at different time. This is due to the random starting point of the trace file and we can therefore note that the video streams are not synchronized for different users and bit rate variations might therefore occur differently for them.

Table 4.5: Contribution of burst losses (in percents) of length n for 10% packet loss.

<table>
<thead>
<tr>
<th>n (packets)</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>8</th>
<th>9</th>
<th>10</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>10% loss rate</strong></td>
<td>MIMO</td>
<td>61.5</td>
<td>17.3</td>
<td>6.7</td>
<td>4.8</td>
<td>4.8</td>
<td>0</td>
<td>1.92</td>
<td>0</td>
<td>0.96</td>
</tr>
<tr>
<td></td>
<td>SIMO</td>
<td>61.9</td>
<td>17.1</td>
<td>7.62</td>
<td>3.81</td>
<td>1.9</td>
<td>0.95</td>
<td>0</td>
<td>1.9</td>
<td>0</td>
</tr>
<tr>
<td><strong>n (packets)</strong></td>
<td>11</td>
<td>12</td>
<td>13</td>
<td>14</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>10% loss rate</strong></td>
<td>MIMO</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0.96</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIMO</td>
<td>0</td>
<td>0.95</td>
<td>0</td>
<td>0.95</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
When the loss rate increases again (up to 10%), the burstiness of the loss pattern becomes even more important. It can be observed in Table 4.5 that the contribution from burst losses of length one packet decreases a little bit more and bigger loss run lengths (such as 12 or 14 packets) can be remarked.

Furthermore, the contribution from loss run lengths exhibits more or less the same values for the case of using MIMO than for the case of using SIMO, what makes the pattern very similar.

We have to note here that some simulations were carried with a video trace file having a mean of 200 kbps, in order to see if a different bit rate gave a different pattern. We observed however that the only difference was that we needed to increase the load of the system to achieve similar loss rates, but the patterns obtained were very similar.

4.3.3. Conclusion

Taking everything into account, we could conclude that, as we had expected, the packet loss pattern from a video communication over a LTE network presents a bursty distribution. Indeed, it could be observed that when a certain packet is lost, the following packets tend to be lost as well and that the bigger the loss rate, the bigger the length of the burst losses.

Finally, we were also able to appreciate that the system using MIMO technology and the system using SIMO technology give more or less the same packet loss pattern, with a similar distribution for the same loss rate.
In order to make an evaluation of the video quality degradation from data packet losses, the burstiness of the packet loss pattern depicted in the previous section needs to be characterized with simple mathematical models.

This section reviews the two mathematical models used to produce typical packet loss patterns, which are able to capture the burstiness of the patterns: the well known Gilbert model and a logarithmic model. In the first place, we carry out a description of both models and in the second place the validity of the models are verified and a comparison between them is presented.

**5.1. Gilbert Model**

The Gilbert Model, analyzed by Sanneck et al. [15] as a tool to capture packet loss burstiness based on loss run lengths, is a two-state Markov model with one “good” state, that represents the state of successful packet arrival, and one “bad” state, that represents the state of packet being lost.

We defined for this model the random variable $X$ as: $X = 0$: *no packet lost*, $X = 1$: *a packet is lost*. Therefore, the state probabilities for the “good” state and the “bad” state are respectively $P(X = 0)$ and $P(X = 1)$.

In addition to that, at each packet interval, the model changes between both states according to the transition probabilities:

- $p_{01} = P(X = 1 | X = 0)$: probability of going from “good” state to “bad” state
- $p_{10} = P(X = 0 | X = 1)$: probability of going from “bad” state to “good” state

It can thus be observed that in the Gilbert model the next event is only dependent in the previous event, i.e. the probability that in the next event a packet is lost or a packet is successfully received only depends in the previous state. The state diagram of the model is shown in Figure 5.1:
Transition and state probabilities can be expressed in matrix form:

\[
\begin{pmatrix}
1-p_{01} & p_{10} \\
p_{01} & 1-p_{10}
\end{pmatrix}
\begin{pmatrix}
P(X = 0) \\
P(X = 1)
\end{pmatrix}
= 
\begin{pmatrix}
P(X = 0) \\
P(X = 1)
\end{pmatrix}
\]  \tag{1}

From the previous relationship (1), and knowing that \( P(X = 0) + P(X = 1) = 1 \) we can calculate the state probabilities as:

\[
P(X = 0) = \frac{p_{10}}{p_{01} + p_{10}} \quad \text{and} \quad P(X = 1) = \frac{p_{01}}{p_{01} + p_{10}}
\]  \tag{2}

If we define at this point:

- \( o_k \) as the number of burst losses of length \( k \)
- \( a \) as the total number of sent packets
- \( d = \sum_{k=1}^{\infty} k \cdot o_k \) as the total number of lost packets

The values for the transition probabilities can be obtained as:

\[
p_{01} = \frac{\sum_{k=1}^{\infty} o_k}{a}
\]  \tag{3}

\[
1-p_{10} = \frac{\sum_{k=1}^{\infty} (k-1) \cdot o_k}{d-1}
\]  \tag{4}

Besides, we also define another random variable \( Y \) which describes the burst loss length, i.e. describes the distribution of loss run lengths with to the burst loss events. Thus, the probability of having a burst loss length of \( k \) packets is:
\[ P(Y = k) = (1 - p_{10})^{k-1} \cdot p_{10}, \ k > 0 \] (5)

It is seen that the probability \( P(Y = k) \) is exponentially decreasing with \( k \) and that the lengths of burst losses given by the Gilbert model are geometrically distributed. The implementation of the Gilbert model in Matlab code is shown in Appendix A.

### 5.2. Logarithmic Model

This Logarithmic model was proposed by Carvalho et al. in [16] to model packet losses in a particular IEEE 802.11g wireless network scenario. Here we employ it to represent the packet loss behavior for video communications in the LTE network.

In order to characterize the loss patterns, this model uses a logarithmic series distribution. If we use again the previously defined random variable \( Y \), being the burst loss length, the probability of a burst loss length of \( k \) packets is in this case:

\[ P(Y = k) = \frac{h \cdot \theta^k}{k}, \ k > 0 \] (6)

where \( \theta \) is a distribution parameter and

\[ h = -\frac{1}{\ln(1 - \theta)}. \] (7)

As it is explained in [16], \( \theta \) needs to be estimated from the observed data so we determine the maximum likelihood estimation of \( \theta \), i.e. \( \hat{\theta} \). This is done by maximizing the following equation:

\[ L = \prod_{i=1}^{N} \frac{h \cdot \theta^{k_i}}{k_i} = \frac{h^N \cdot \prod_{i=1}^{N} k_i}{\prod_{i=1}^{N} k_i} \] (8)

where \( k_i, \ i = 1,2,...N \) are the observations of the different lengths of the burst losses. Therefore, if we take the natural logarithmic of both sides of (8) we get:

\[ \lambda = \ln(L) = N \cdot \ln(h) + \sum_{i=1}^{N} k_i \cdot \ln(\theta) - \ln(\prod_{i=1}^{N} k_i) \] (9)

We calculate now the partial derivation of \( \lambda \) with respect to \( \hat{\theta} \):

\[ \frac{\partial \lambda}{\partial \theta} = \frac{\partial}{\partial \theta} (N \cdot \ln(h)) + \frac{\partial}{\partial \theta} (\sum_{i=1}^{N} k_i \cdot \ln(\hat{\theta})) - \frac{\partial}{\partial \theta} (\ln(\prod_{i=1}^{N} k_i)) \] (10)
By setting the partial derivation of \( \lambda \) with respect to \( \theta \) (10) equal to zero, we obtain:

$$\frac{\partial \lambda}{\partial \theta} = -\frac{Nh}{(1-\theta)} + \frac{1}{\theta} \sum_{i=1}^{N} k_i = 0 .$$

(14)

Finally, rearranging the above equation (14) and using again the definitions \( o_k \) number of burst losses of length \( k \) and \( d = \sum_{k=1}^{\infty} k \cdot o_k \) total number of lost packets, we get:

$$\frac{h \cdot \bar{\theta}}{(1-\bar{\theta})} = \frac{1}{N} \sum_{i=1}^{N} k_i = \frac{d}{\sum_{i=1}^{\infty} o_i} .$$

(15)

The maximum likelihood estimation of \( \theta \), i.e. \( \bar{\theta} \), can be calculated from equation (15) and the probability \( P(Y = k) \) can therefore be estimated. The implementation of the Logarithmic model in Matlab code is shown in Appendix A.

5.3. Verification

The validity of both models is verified by obtaining statistics from the simulation data and comparing them to the statistics predicted by the models. In this case, we compare the probability of having burst losses of various lengths, i.e. \( P(Y = k) \), from the simulations with the probabilities obtained from the different models.

As we have already explained, to compute the desired probability for the Gilbert model we use the equation (5), having previously calculated the transition probabilities with equations (3) and (4). To calculate this probability for the Logarithmic model, we use the equation (6) having previously calculated the maximum likelihood estimation of the distribution parameter with (15). We therefore used the parameters \( a, d \) and \( o_k \) from the simulation as input data for the models.
5.3.1. First approach

In the first place, we conducted a graphic comparison between the distribution of burst losses derived from the models and the distribution obtained from the simulations. In Figures 5.2 and 5.3 this comparison is illustrated for two specific simulations with different loss rates: 5 and 10% respectively.

Figure 5.2: Comparison of the probabilities of loss run lengths from the models and a simulation with a loss rate of 5%.
Figure 5.3: Comparison of the probabilities of loss run lengths from the models and a simulation with a loss rate of 10%.

Since each figure illustrates the case of a single simulation, they should only be used as a hint of the comparison between the models and the simulation data. However, we can appreciate that, even if both models seem to be able to characterize the distribution of burst losses, the Logarithmic model may be more accurate than the Gilbert model. Indeed, it can be observed in both cases that the Logarithmic model curve is closer to the simulation curve than the Gilbert model curve. What is more, we can also perceive that the bigger accuracy of the Logarithmic model is more sensible for smaller lengths of the burst losses.

5.3.2. Correlation and MSE

In order to certify the statements described above, we made use of two comparative parameters: the correlation and the mean square error (MSE). Both parameters helped to carry out the verification and the comparison between the models and the simulation data.

Correlation Coefficient

The correlation is one of the most common and most useful statistics. The correlation coefficient (also known as Pearson’s product-moment correlation coefficient) is a single number that describes the degree of relationship between two variables which may take any value between -1.0 and +1.0 (1 in the case of an increasing linear relationship, −1 in the case of a decreasing linear relationship, and some value in between in all other cases) [17].

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Indeed, if we have two random variables $X$ and $Y$ with $n$ measurements written as $x_i$ and $y_i$ (with $i = 1, 2, \ldots, n$), the correlation coefficient $\rho$ can be obtained as:

$$
\rho = \frac{\sum_{i=1}^{n} x_i \cdot y_i - \frac{1}{n} \sum_{i=1}^{n} x_i \cdot \sum_{i=1}^{n} y_i}{\sqrt{\sum_{i=1}^{n} x_i^2 - \frac{1}{n} \left( \sum_{i=1}^{n} x_i \right)^2} \cdot \sqrt{\sum_{i=1}^{n} y_i^2 - \frac{1}{n} \left( \sum_{i=1}^{n} y_i \right)^2}}
$$

In consequence, the correlation coefficient between the probabilities of having burst losses of various lengths from the simulation data and from each one of the models was calculated.

**Mean Square Error**

Apart from the correlation, we also used another comparative parameter, the mean square error (MSE) [17]. It is defined as the difference between some set of observations and the response predicted by the model and it is used to determine how well the model fit the observations. The MSE is defined as:

$$
MSE = \frac{1}{n} \sum_{i=1}^{n} \varepsilon_i^2
$$

where $\varepsilon_i$, $i = 1, 2, \ldots, n$ (i.e. $n$ measurements) is the difference between the observed data and the model.

However, the MSE by itself does not permit to draw important conclusions. Thus, two or more statistical models can be compared using their MSEs. In this case, the model with smallest MSE is seen as the best explaining a set of observations.

For this reason, the MSE between the simulation data and the models were calculated and compared. In Table 5.1 the results for the correlation coefficient and the MSE are summarized.

<table>
<thead>
<tr>
<th>Comparison</th>
<th>Correlation coefficient, $\rho$</th>
<th>MSE</th>
</tr>
</thead>
<tbody>
<tr>
<td>Between simulation data and Gilbert model</td>
<td>0.935</td>
<td>0.0041</td>
</tr>
<tr>
<td>Between simulation data and Logarithmic model</td>
<td>0.975</td>
<td>0.0022</td>
</tr>
</tbody>
</table>
5.3.3. Conclusion

Firstly, it can be observed in Table 5.1 that both models have a big correlation coefficient, i.e. the distribution of loss run lengths from the simulations and the distributions given by the models have a strong relationship. Yet, the correlation between the simulation data and the Logarithmic model is greater than the correlation between the simulation data and the Gilbert model.

Secondly, we can also appreciate that the MSE between the simulation data and the Logarithmic model is smaller than the MSE between the simulation data and the Gilbert model; i.e. the former explains better the simulation data than the latter.

In short, the results illustrate that the Logarithmic model is more accurate and fits better the simulation data than the Gilbert model, especially for smaller lengths of burst losses.
6. Video Quality Evaluation

As section 4 illustrates, the losses in video communication over a LTE network are bursty. Moreover, [11] analyzes and explains that the actual packet loss pattern, and in particular the burst length, is important and has a significant effect on the resulting degradation of the video. In fact, this effect is particularly pronounced for low bit rate video since, in this case, each frame may be coded into a single packet. Therefore, a burst loss of several packets may lead to the loss of several frames and the effect will be appreciable. However, for the case of high bit rate, each frame can be coded into 10 packets - hence, much longer burst losses are required to have similar effects.

The objective of this section is to evaluate the effect of the bursty packet loss patterns on real video clips. To do so, VideoBatchCoder (VBC) is used. This program enables to analyze the degradation caused by some packet losses on real video clips, making use of the Perceptual Evaluation of Video Quality (PEVQ) score. Therefore, we analyzed with VBC the video quality degradation from the simulation packet loss patterns and from the modeled ones.

In this section, we first carry out a description of the VBC program and the PEVQ and we explain the different settings and parameters applied to the program. Secondly, we analyze the results obtained and we work out a comparison between the degradation caused by the simulated patterns and the degradation caused by the modeled patterns. Finally, we draw some conclusions.

6.1. VideoBatchCoder (VBC)

6.1.1. Description

VBC is a program with which it is possible to carry out different media quality measurements. By introducing a variety of combinations of input files, encoders and coding parameters, you can produce multiple yuv, avi, wav bitstream files, etc. For instance, when an input file is an avi file, the program encodes the frames into RTP packets; and depending on the other input files and parameters, e.g. an input packet loss rate, the program implements a simulation where some packet will be lost according to this loss rate, obtaining a resulting output avi file.
As Figure 6.1 illustrates, in our case the input files that we used are a real video clip (i.e. an avi file) and a file containing a list with the lost packets, i.e. with the packet loss pattern. By combining both input files, VBC returns the degraded version of the original video clip. It can then perform the PEVQ algorithm by comparing the original and the degraded video file, and therefore calculate the PEVQ score.

**Perceptual Evaluation of Video Quality (PEVQ)**

PEVQ [18] is a ITU standardized, full-reference, end-to-end measurement algorithm to analyze the picture quality of a specific video. The measurement is based on signal analysis: the degraded video signal output is analyzed by comparison to the undistorted original reference video signal on a perceptual basis. Based on the approach to model the human visual system, PEVQ can detect anomalies and quantify them by a number of key performance indicators (KPIs).

The most important KPI is the PEVQ score, which specifies the video quality by means of a 5-point mean opinion score (MOS) [19] where 1 corresponds to bad quality and 5 to excellent quality (Table 6.1). However, we may obtain sometimes a PEVQ score smaller than 1, which indicates that the quality is below the worst case of the MOS scale. The PEVQ score is based on a multitude of perceptually motivated parameters.
Table 6.1: PEVQ score scale. Adapted from [19].

<table>
<thead>
<tr>
<th>PEVQ score</th>
<th>Quality</th>
<th>Degradation</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>Excellent</td>
<td>Imperceptible</td>
</tr>
<tr>
<td>4</td>
<td>Good</td>
<td>Perceptible but not annoying</td>
</tr>
<tr>
<td>3</td>
<td>Fair</td>
<td>Slightly annoying</td>
</tr>
<tr>
<td>2</td>
<td>Poor</td>
<td>Annoying</td>
</tr>
<tr>
<td>1</td>
<td>Bad</td>
<td>Very annoying</td>
</tr>
</tbody>
</table>

Figure 6.2 outlines the relationship between the PEVQ score and the packet loss rate. Clearly, the bigger the packet loss rate the bigger the quality degradation of a specific video. Therefore, it can be observed that when the packet losses increase the PEVQ score decreases exponentially.

![Figure 6.2: PEVQ score vs. packet loss rate.](image)

6.1.2. Settings

In order to conduct the video quality evaluation, different settings were applied to the VBC program. First of all, different types of input avi files were used. To be able to obtain more widespread results, each video used had different features. Indeed, the clips we used were:

- A video containing part of the movie *Pirates of the Caribbean* (henceforth called *POTC* video). This video has a lot of movement and the frames are very different from each other. However, the video is generally darkish and few scenes are bright.
• A video containing part of a football match (henceforth called Football video). In this case, the video has a lot of color all over the screen (due to the green of the football field) and a lot of brightness. Moreover, the video has also a lot of movement.

• A video containing part of a television newscast (actually, a journalist in a set giving the news) (henceforth called News video). This video has also a lot of color and brightness but in this case there is no much movement.

Apart from the video files, another input file was a text file that included the packets that were lost, as we have already explained. In fact, for each evaluation this file was extracted from the simulation patterns and from the modeled patterns for each model. In Appendix B, it is shown in Matlab code how the patterns from the models are produced.

Furthermore, VBC permits to set several input parameters. In our case, for each evaluation we defined a video bit rate of 800 kbps. The main settings and parameters used are summarized in Table 6.2.

Table 6.2: VBC settings.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video bit rate</td>
<td>800 kbps</td>
</tr>
<tr>
<td>Codec</td>
<td>H.264</td>
</tr>
<tr>
<td>Frame format resolution</td>
<td>QVGA (320 x 240)</td>
</tr>
<tr>
<td>Frame rate</td>
<td>25 fps</td>
</tr>
</tbody>
</table>

### 6.2. Results

After applying the correct settings and parameters, we conducted a series of evaluation of the video quality. To do so, we combined the three input video files with the different packet loss patterns files from the simulations and from the corresponding modeled patterns for several packet loss rates (1, 3, 5, 7.5 and 10%).

Thus, we were able to gather the PEVQ score for different loss rates and compare it between the score obtained for the simulation patterns and the one obtained for the patterns given by each model. In Figures 6.3, 6.4 and 6.5 this comparison is illustrated for the POTC video, the Football video and the News video respectively.
Figure 6.3: Comparison of the PEVQ score for the simulation and for the Gilbert and the Logarithmic models. *POTC* video.

Figure 6.4: Comparison of the PEVQ score for the simulation and for the Gilbert and the Logarithmic models. *Football* video.
To be able to carry out a better comparison, the correlation coefficient was used again and the correlation between the PEVQ score from the simulation patterns and the PEVQ score from the patterns modeled with the Gilbert model was calculated, as well as the correlation between the PEVQ score from the simulation patterns and the PEVQ score from the patterns modeled with the Logarithmic model. In Table 6.3, the results for the correlation for each input video clip are summarized.

Table 6.3: Correlation coefficient between PEVQ scores for simulation data and models.

<table>
<thead>
<tr>
<th>Comparison</th>
<th>POTC video</th>
<th>Football video</th>
<th>News video</th>
</tr>
</thead>
<tbody>
<tr>
<td>Between simulation data and Gilbert model</td>
<td>0.841</td>
<td>0.825</td>
<td>0.738</td>
</tr>
<tr>
<td>Between simulation data and Logarithmic model</td>
<td>0.975</td>
<td>0.863</td>
<td>0.916</td>
</tr>
</tbody>
</table>
6.3. Conclusions

First of all, from Figures 6.3, 6.4 and 6.5 we were able to draw some conclusions. Indeed, we could appreciate that the relationship between the PEVQ score and the packet loss rate was as expected: when the packet loss rate increases, the PEVQ score decreases. Moreover, we can note that when the packet loss is 0, the PEVQ score is slightly lower than 5 due to encoding reasons that are not explained here.

Apart from that, we could observe that due to the varied characteristics of the input video clips, the patterns have different influence in each one of them and for a same loss rate the PEVQ score may be different for each clip:

- For the POTC video, the degradation is not so important for a low loss rate (1%). However, when the loss rate increases a bit (up to 3%), the PEVQ score drops more than one point and the degradation becomes appreciable. We could therefore conclude that this kind of video admits a certain loss rate but is sensible to an increase of the packet loss rate.

- For the Football video, the PEVQ score is low even with low loss rate. Therefore, for videos with these features the quality degradation is very sensible to packet losses and, to watch this kind of videos with a reasonable quality, the packet loss rate should be close to zero.

- For the News video, we could observe that the PEVQ score maintains a big value even when the loss rate increases. Thus, we could affirm that the quality degradation for this kind of videos is less sensible to packet loss rate.

Moreover, as expected also from section 5, the figures show that the Logarithmic model seems to be more accurate for the three video clips, since the PEVQ score obtained for the degradation from the patterns modeled by the Logarithmic model is closer to the PEVQ score for the simulation patterns than the PEVQ score obtained for the degradation from the patterns modeled by the Gilbert model.

This claim is proved by the results shown in Table 6.3. It can be observed that for the three video clips the correlation coefficient between the PEVQ score for the simulation patterns and for the Logarithmic patterns is greater than the correlation coefficient between the PEVQ score for the simulation and the Gilbert patterns. In fact, for the POTC video and the News video, the correlation between the simulation and the Logarithmic modeled patterns is big. For the Football video, even if this correlation is a little bit lower (surely due to its characteristics), it remains still bigger.

To sum up, we could conclude that the video quality degradation in a LTE network is better characterized by the packet loss patterns modeled by the Logarithmic model than by the patterns modeled by the Gilbert model – which is in concordance with the fact, verified in section 5, that the Logarithmic model is more accurate.
7. Conclusions

Throughout this master thesis, the behavior of the LTE network with respect to packet losses at application level, where a packet is regarded as lost if arriving to late, has been analyzed thanks to a LTE simulator. Indeed, the packet loss patterns obtained from the simulations have been analyzed and described. Afterwards, an evaluation of the video quality degradation due to these packet loss patterns has been carried out.

As we expected by looking into other studies about packet transmissions over other types of real networks, typical packet loss patterns in a LTE network are bursty. In fact, for low loss rates the burst losses of length 1 (i.e. single packet losses) are the most important loss run lengths. However, when the loss rate increases the burst losses of bigger lengths become more important and their occurrence increases with the increase of the packet loss rate. Therefore, we could conclude that typical packet loss patterns in LTE have a bursty distribution of the losses, i.e. when a specific packet is lost there is high probability that the following packet will be lost as well.

Thus, in order to accomplish the quality evaluation, the packet loss patterns identified needed to be modeled. Two mathematical models, which a priori could characterize the burstiness of a specific loss pattern, have been identified. Indeed, the validity of both models has been verified, i.e. both models were able to model the distribution of burst losses. However, a comparison between them has showed that the Logarithmic model is more accurate than the Gilbert model.

Finally, the video quality degradation has been analyzed taking into account the simulation patterns, the patterns modeled with the Logarithmic model and the patterns modeled with the Gilbert model. Three different input video files, with distinctive characteristics, have been used and we could conclude that the packet losses have different effect on the original video clips depending on their features. Moreover, we could observe that the degradation caused by the patterns modeled with the Logarithmic model is closer to the degradation caused by the simulation patterns than the degradation caused by the patterns modeled with the Gilbert model.

As a result, the video quality degradation has been successfully evaluated and it can be modeled accurately by the Logarithmic model analyzed in this work, enabling to construct degraded video files for use in for example subjective testing.
8. Future Work

LTE is the radio network that will be used in the near future, which means that there is much research going on and many issues that remain to be solved. In fact, as we have explained before, in this thesis we have run the LTE simulations for a specific mobile TV user: a real-time video user. Thus, the streaming video user, where the user selects clips of interest for viewing and represents rather on-demand TV, has been left apart.

Therefore, further studies could be carried out in order to analyze this streaming video user. Instead of a loss pattern, a “delay” pattern could be depicted where the occurrence of the rebuffering functionality is described to be able to know when the user would have to wait.

Furthermore, we do not know what kind of user is going to be the main user in the LTE network, since by now both types of video communication have approximately the same usage rate. Therefore, future work could be done in order to learn what kind of usage case will take advantage so that more investments and efforts could be assigned to investigate it.

Apart from that, as regards the real-time video user, the packet loss patterns have been studied focusing on the loss run lengths, where the packet loss burstiness was analyzed only as consecutive losses. However, another approach would be to consider a burst as a longer period of high loss density, beginning and ending with a loss during which the number of consecutive received packets is less than e.g. 5 packets. In this case, if a specific packet is lost and the following two packets are received but the third one is also lost, it would not count as two burst losses of length 1, but rather a single burst loss of length two. For this new criterion, other mathematical models should be identified to be able to characterize the new burstiness definition.

Finally, we have to note that at the beginning the objectives of the master thesis included the LTE network and the HSPA network to be evaluated. However, due to time restrictions, the video quality degradation has been investigated only in the LTE network. Hence, the evaluation of video quality degradation from data packet losses in a HSPA network could be subject of further studies.
References


Appendix A: Implementation of Mathematical Models in Matlab Code

Implementation of the Gilbert Model

```matlab
function PYkmod = GilbertModel(lossPattern)

p01 = sum(lossPattern.BurstLengths)/length(lossPattern.Pattern);

s = 0;
for i = 1:length(lossPattern.BurstLengths)
    s = s + (i-1)*lossPattern.BurstLengths(i);
end

nrLostPackets = 0;
for i = 1:length(lossPattern.BurstLengths)
    nrLostPackets = nrLostPackets + (i)*lossPattern.BurstLengths(i);
end

p11 = s/(nrLostPackets - 1);
p10 = 1 - p11;
P1 = p01/(p01 + p10);
P0 = p10/(p01 + p10);
PYkmod = zeros(1,length(lossPattern.BurstLengths));
for i = 1:length(lossPattern.BurstLengths)
    PYkmod(i) = (p11^(i-1))*p10;
end
end
```

Implementation of the Logarithmic Model

```matlab
function PYkmod = LogModel(lossPattern)

nrLostPackets = 0;
for i = 1:length(lossPattern.BurstLengths)
    nrLostPackets = nrLostPackets + (i)*lossPattern.BurstLengths(i);
end

mean = nrLostPackets/sum(lossPattern.BurstLengths);
```
teta = fzero(@(x) myfun(x,mean), 0.5);

PYkmod = zeros(1,length(lossPattern.BurstLengths));

for i = 1:length(lossPattern.BurstLengths)
    PYkmod(i) = ((-1/log(1-teta))*teta^i)/i;
end

where *myfun* is a Matlab function containing the desired mathematical function:

```matlab
function f = myfun(x,c)

    f = ((-1/log(1-x))*x)/(1-x) - c;

end
```
Appendix B: Implementation of the Modeled Patterns

Implementation of the Pattern from the Gilbert Model

function GilbertPattern = createGilbPattern(lossPattern)

gilbertPattern = zeros(1,length(lossPattern.Pattern));

p01 = sum(lossPattern.BurstLengths)/length(lossPattern.Pattern);

s = 0;
for i = 1:length(lossPattern.BurstLengths)
    s = s + (i-1)*lossPattern.BurstLengths(i);
end

nrLostPackets = 0;
for i = 1:length(lossPattern.BurstLengths)
    nrLostPackets = nrLostPackets + (i)*lossPattern.BurstLengths(i);
end

p11 = s/(nrLostPackets - 1);
p10 = 1 - p11;

gilbertPattern(1) = 0; %we assume the first packet has been successfully received

for i = 2:length(gilbertPattern)
    r = rand(1);
    if (gilbertPattern(i-1) == 0)
        if (r <= p01)
            gilbertPattern(i) = 1;
        else
            gilbertPattern(i) = 0;
        end
    end
    if (gilbertPattern(i-1) == 1)
        if (r <= p10)
            gilbertPattern(i) = 0;
        else
            gilbertPattern(i) = 1;
        end
    end
end

GilbertPattern = testPattern(gilbertPattern)
end
**Implementation of the Pattern from the Logarithmic Model**

```matlab
function LogPattern = createLogPattern(lossPattern)

logPattern = zeros(1,length(lossPattern.Pattern));

PYkmod = LogModel(lossPattern);
PYkmodRel = PYkmod * (lossPattern.PercentLosses/100);

okmod = PYkmod*sum(lossPattern.BurstLengths);

logPattern(1) = 0; %we assume the first packet has been succesfully received

for i = 2:length(logPattern)
    if (sum(logPattern(1:i)) >= lossPattern.NrLostPackets)
        break
    end
    if (logPattern(i) == 0)
        if (logPattern(i-1) == 1)
            logPattern(i) = 0;
        else
            r1 = rand(1);
            if (r1 > (lossPattern.PercentLosses/100))
                logPattern(i) = 0;
            else
                if (r1 <= PYkmodRel(1))
                    logPattern(i) = 1;
                else
                    for j = 2:length(PYkmodRel)
                        if (j > (length(logPattern) - i + 1))
                            logPattern(i:length(logPattern)) = 1;
                        else
                            logPattern(i:(i+(j-1))) = 1;
                        end
                    end
                end
            end
        end
    end
end

LogPattern = testPattern(logPattern)
end
```

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In both cases, the function `testPattern` is implemented as:

```matlab
function information = testPattern(pattern)

nrLostPackets = sum(pattern);

lossRate = (nrLostPackets/length(pattern))*100;

burstLengths = burstLength (pattern);

lostPacketsNr = obtainLostPckNr(pattern);

information = struct('Pattern', pattern, 'PercentLosses', lossRate, 'BurstLengths', burstLengths, 'NrLostPackets', nrLostPackets, 'LostPacketsNr', lostPacketsNr);
end
```

Moreover, to create a text file containing a list with the packets that are lost from a specific pattern, we used:

```matlab
function createtxt(fileName, Pattern)

fid = fopen(fileName,'w');
data = Pattern.LostPacketsNr;

fprintf(fid, '%u\r\n', data);
fclose(fid);
end
```