Simulation Estimation of Network and Quality Characteristics in Video Transmission over LTE Network

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Abstract – **In this paper are presented the main technical features of LTE standard. An analysis of LTE physical layer is also given and simulation of video transmission with different resolution and frame rates was conducted.**

***Keywords* – LTE, CQI, video transmission, jitter.**

I. Introduction

The era of new wireless communications is upon us. Telecommunications technologies are evolving at an unbelievable pace, driven by innovation, speed, quality, convenience, and cost. Long-Term Evolution (LTE) is the standard recently specified by the 3GPP on the way towards fourth-generation mobile. LTE leads a big technological improvement as compared with the previous 3G standard.

LTE is designed to increase data rates and cell edge bitrates, improve spectrum efficiency (unicast as well as broadcast) and allow spectrum flexibility (1.25, 2.5, 5, 10, 15 and 20 MHz) for flexible radio planning. LTE has also to reduce packet latency, the main restriction for real-time services, such as VoIP, video-conferencing, stream video, etc., reduce radio access network cost as well as cost-effective migration from earlier 3GPP releases and simplify its network to a flat all-IP packet-based network architecture where all the user plane radio functionalities are terminated at the enhanced Node B (eNB). [1]

II. LTE Version 8 Main Features

The radio access network of LTE is called E-UTRAN and one of its main features is that all services, including real-time, will be supported over shared packet channels. This approach will achieve increased spectral efficiency which will turn into higher system capacity with respect to current UMTS and HSPA. An important consequence of using packet access for all services is the better integration among all multimedia services and among wireless and fixed services. [2]

The main philosophy behind LTE is minimizing the number of nodes. Therefore the developers opted for a single-node architecture. The new base station is more complicated than the Node B in WCDMA/HSPA radio access networks, and is consequently called eNB (Enhanced Node B). The eNBs have all necessary functionalities for LTE radio access network including the functions related to radio resource management. The new core network is a radical evolution of the one of third generation systems and it only covers the packet-switched domain. Therefore it has a new name: Evolved Packet Core. Following the same philosophy as for the E-UTRAN, the number of nodes is reduced. Evolved Packet Core (EPC) divides user data flows into the control and the data planes. A specific node is defined for each plane plus the generic gateway that connects the LTE network to the internet and other systems. The EPC comprises several functional entities. [3]

The overall structure of LTE is shown in Fig. 1, where: the Mobility Management Entity (MME) is responsible for the control plane functions related to subscriber and session management; The Serving Gateway is the anchor point of the packet data interface towards E-UTRAN. Moreover, it acts as the routing node towards other 3GPP technologies; The Packet Data Network (PDN Gateway) is the termination point for sessions towards the external packet data network. It is also the router to the Internet; The Policy and Charging Rules Function (PCRF) controls the tariff making and the IP Multimedia Subsystem (IMS) configuration of each user. [4]

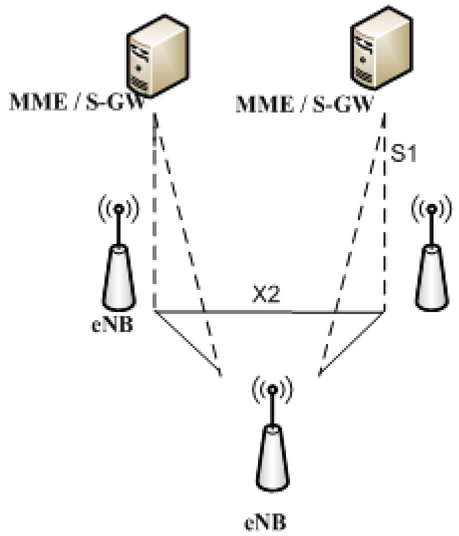

Fig. 1. LTE architecture

III. Channel quality parameters

In modern wireless networks, the signal quality in wireless channel is estimated based on the channel quality measurements. The measurement results are used to select appropriate modulation and coding scheme for each transmission. In the downlink of Long Term Evolution (LTE) systems, feedback and processing delays cause a mismatch

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between the current channel state and the Channel Quality Information (CQI) at the base station. This CQI aging leads to inaccurate channel adaptation and can, thus, highly degrade the cell capacity.[5, 6]

The Channel Quality Indicator (CQI) contains information sent from a UE to the eNode-B to indicate a suitable downlink transmission data rate, i.e., a Modulation and Coding Scheme (MCS) value. CQI is a 4-bit integer and is based on the observed signal-to- interference-plus-noise ratio (SINR) at the UE. The CQI estimation process takes into account the UE capability such as the number of antennas and the type of receiver used for detection. This is important since for the same SINR value the MCS level that can be supported by a UE depends on these various UE capabilities, which needs to be taken into account in order for the eNode-B to select an optimum MCS level for the transmission. The CQI reported values are used by the eNode-B for downlink scheduling and link adaptation, which are important features of LTE.

The supported CQI indices and their interpretations are given in Table 1. In total, there are 16 CQI values, which require a 4-bit CQI feedback. In Table 1, the efficiency for a given CQI index is calculated as:

$$E = Q_m \times R \tag{1}$$

where Q_m is the number of bits in the modulation constellation.

TABLE I
CQI INDEXES

CQI index	Modulation	Code rate R	Efficiency
0	-		
1	QPSK	78	0,1523
2	QPSK	120	0,2344
3	QPSK	193	0,3770
4	QPSK	308	0,6016
5	QPSK	449	0,8770
6	QPSK	602	1,1758
7	16QAM	378	1,4766
8	16QAM	490	1,9141
9	16QAM	616	2,4063
10	64QAM	466	2,7305
11	64QAM	567	3,3223
12	64QAM	666	3,9023
13	64QAM	772	4,5234
14	64QAM	873	5,1152
15	64QAM	948	5,5547

Based on the estimated effective SINR, the UE picks the CQI index that indicates the highest MCS level (modulation and code rate) that can be supported with a 10% BLER on the first H-ARQ transmission. The CQI feedback is used by the eNode-B to select an optimum PDSCH transport block with a combination of modulation scheme and transport block size corresponding to the CQI index that could be received with target block error probability after the first H-ARQ transmission. While this target block error probability is left open as an implementation choice, typical values are in the range of 10-25%. It should be noted that the target BLER of the transmission is not the same as the BLER of 10% based on which the CQI is computed. Thus, the eNode-B needs to take this into account while selecting the optimum transport block size. If the achieved block error rate is not equal to the target value based on the H-ARQ ACK/NAK ratio, then a fudge factor can be added to the CQI to ensure that the selection of the block size based on the CQI leads to the desired target block error rate. A positive fudge factor implies a more aggressive transport block size selection, whereas a negative fudge factor implies a more conservative transport block size selection. [6]

Figure 2 represents block error rate (BLER) in function of signal-to-noise ratio (SNR) by various CQI indexes.

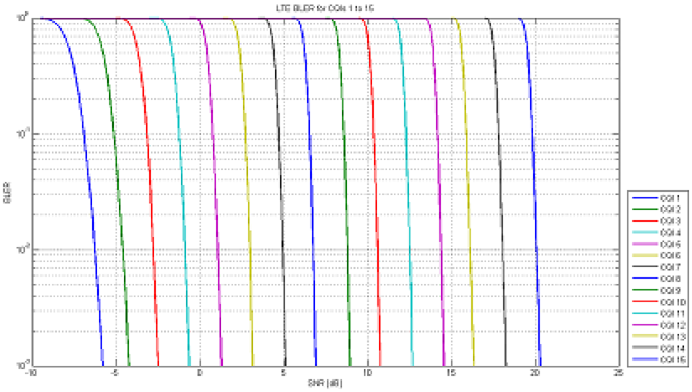

Fig. 2. BLER in various CQI indexes

IV. SIMULATION RESULTS

For transmission simulation we use NS-2 network simulator with two additional modules. The first one is EvalVid – a complete framework and tool-set for evaluation of the quality of video transmitted over a real or simulated communication network. Besides measuring QoS parameters of the underlying network, like loss rates, delays, and jitter, it supports also a subjective video quality evaluation of the received video based on the frame-by-frame PSNR calculation. The tool-set has a modular construction, making it possible to exchange both the network and the codec. The processing of the data takes place in 3 stages. The first stage requires the timestamps from both sides and the packet types. The results of this stage are the frame-type based loss rates and the inter-packet times. Furthermore the erroneous video file from the receiver side is reconstructed using the original encoded video file and the packet loss information. This video can now be decoded yielding the raw video frames which would be displayed to the user. At this point a common problem of video quality evaluation comes up. Video quality metrics always require the comparison of the displayed (possibly distorted) frame with the corresponding original frame. In the case of completely lost frames, the required synchronization cannot be kept up. The second stage of the processing provides a solution to this problem. Based on the loss information, frame synchronization is recovered by inserting the last displayed frame for every lost frame. This makes further quality assessment possible. The thus fixed raw video file and the original raw video file are used in the last stage to obtain the video quality. [7]

The second additional module we use is Long Term Evolution/System Architecture Evolution (LTE/SAE) module, introduced in [10]. The system model includes LTE/SAE network model, traffic model and flow control model. In the network model, some network configuration parameters can easily be changed, for example it can be defined any number of UEs, bandwidth between the network elements and the usage of the optimization features; others cannot be changed so easily due to the limitation of the implemented model, such as eNB and advanced gateway (aGW) number. In the simulated LTE/SAE network, the following network elements are included – 1 server (provide HTTP, FTP and signaling services), 1 aGW (provide HTTP cache, flow control), 1 eNB (provide flow control information) and many UEs. In this module three types of traffic are provided – Conversational, Streaming and Interactive traffic. The Conversational traffic simulates transmission of Voice over IP, Video conferencing calls and Telephony speech. Interactive traffic simulates Web browsing and e-mail reading/sending. The most important traffic type for us is the Streaming traffic. It's one-way traffic and it is used for Audio and Video streaming. The QoS mechanisms provided in the cellular network have to be robust and capable of providing reasonable QoS resolution. More accurate flow controls, better performance the system can get. However, the system's resource, such as CPU and memory, is limited. Balance between the system performance and resource is needed. Hierarchy Virtual Queue (HVQ) flow control provides the good solution to get the high ratio of performance over resource. The basic principle of the virtual queue based flow control is shown in the following figure.

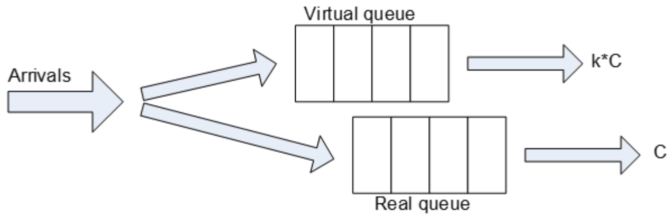

Fig. 3. Virtual queue based flow control

A virtual queue is a fictitious queue with a capacity less than the actually available capacity. The motivation for using the virtual queue is that it provides advance warning of congestion. For each real queue there is a corresponding virtual queue or a set of virtual queues, one for each differently treated traffic class. When packets arrive into the real queue, the virtual queue length is also updated by the length of the received packet. The draining rate of the virtual queue is k*C (k≤1), where C is the draining rate of the real queue. When the virtual queue length upper threshold reaches, the flow control information is send from air interface to the S1 interface. S1 interface will take the corresponding flow control actions. If the flow control information is about UE, the packets belong to the UE are blocked; if the flow control information is about cell, all the packets belong to the cell where the UE locates are blocked; if the flow control information is about the eNB, all packets belong to the eNB where the UE locates are blocked. The blocking is cancelled when the lower threshold is triggered. [8]

Fig. 4 shows the architecture of the simulation. During the simulation study, three video clips with different frame rates, resolution and size are used. The parameters of the clips are shown in Table 2.

Fig. 4. Architecture of the system

TABLE II
PARAMETERS OF TRANSMITTED VIDEO CLIPS

Clip	Resolution	FPS	Duration, min	Size, Mb
A Bit of a Pickle	1920x1080	24	03:45	90,6
I Nub You	1280x720	24	02:58	39,4
Container	352x288	30	00:10	13,5

The most important parameter in wireless video transmission is the jitter. From the conducted simulation and the graph, shown in Fig. 5, it can be seen, that this parameter has a very low value, which is prerequisite for excellent delivery and play in user side.

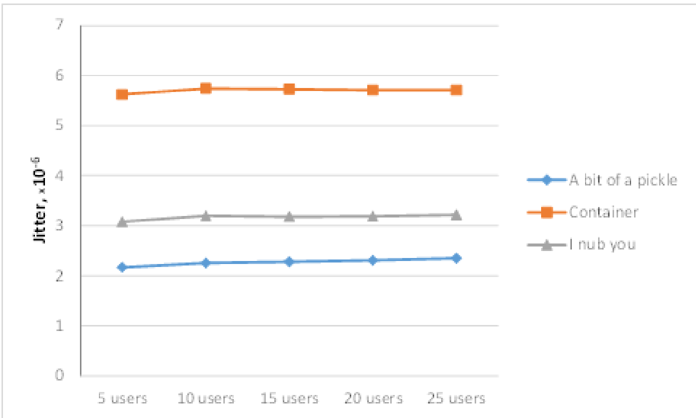

Fig. 5. Jitter of transmitted video clips

V. CONCLUSION

LTE has been designed as a future technology to cope with next user requirements. We have presented a framework for an analytical comparison between the achievable information rate in SC-FDMA and that in OFDMA. Based on the conducted simulation it can be seen that video files with higher resolution (1920x1080) have lower jitter than the video files with resolution (352x288).

Future work will focus on more detailed analysis, design and simulation of transmission of video information with high resolution to more subscribers (UEs). Finally, combining channel codding influence and channel capacity is an emerging area that will be the subject of future research.

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