Evaluation of the Speech QoE in Voice over LTE services

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Abstract **– This work introduces a mathematical relationship linking Quality of Experience (QoE) and Quality of Service (QoS) in Voice over Long Term Evolution (VoLTE) services, validated through an OPNET simulation model. Moreover, a real world test is reported that confirms the simulation validation. Besides its academic interest, knowledge of the relationship can provide Long Term Evolution (LTE) network management a way to predict the QoE offered to VoLTE users, by simply knowing the measurements of the network QoS.**

Keywords – QoE; QoS; User Experience; LTE; VoLTE; OPNET simulation model.

I. INTRODUCTION

In the network management process, it is of interest to the network operator to focus on user satisfaction, by continuously monitoring the user QoE and, if necessary, adjust the QoE by operating on the network QoS.

In order to do this, it is essential to establish a link between QoE perceived by the user and QoS offered by the network. QoS defines the network quality and it is measured by quantitative parameters named Key Performance Indicators (KPI), such as packet loss, delay and jitter [1]-[4]. On the other hand, QoE defines the quality subjectively perceived by the end-user, and gives information on how well the network meets the user's needs. QoE is measured by qualitative parameters, named Key Quality Indicators (KQI), such as "very good", "good", "poor"[5]-[8].

If the relationship between QoE and QoS is known, the network operator may measure the network KPIs and adjust them to obtain satisfactory KQIs for the end-user. So, it is important to make such a relationship known.

The LTE standard assures to deliver high performance and quality IP services, like voice call, but only focuses on the definition of network performance (QoS), without giving its relationship with the QoE [9]-[13].

However, the network operator needs to exploit such a relationship in order to meet the Service Level Agreement (SLA) established with the end-user. By knowing this relationship, the network operator may predict the QoE that can be offered simply based on QoS measurements.

The scope of this paper is thus to mathematically express this relationship in network scenarios delivering VoLTE services.

The paper is organized as follows. Section II introduces the basic mathematics to evaluate the VoLTE QoE. Section III introduces the related works. Section IV introduces proposed relationship between QoE and QoS in VoLTE and Section V validates such a relationship in simulated scenarios.

II. BASIC ANALYTICS FOR THE VOLTE QOE

VoLTE is the voice service delivered in LTE all-IP Packet-Switched domain, in which the network operator provides the service by means of its own network, so that he can manage all phases of the service.

VoLTE calls provisioning is made possible by specific architectural elements composing the so called Internet Multimedia Subsystem (IMS), standardized by Third Generation Partnership Project (3GPP) in [14][15].

In considering the User Plane, some remarks are necessary in understanding the VoLTE procedure. For what concerns the protocol stack, the User Equipment (UE) and the IMS entities that terminate the User Plane must use the Real Time Protocol (RTP) at Application Layer [16]. So, voice streams of the same voice call follow the same RTP flow. However, VoLTE is delivered in a packet switched domain, where IP packets of the same RTP flow can reach the destination by means of different network paths, and consequently with different delays.

This phenomenon is measured by a KPI, also known as network jitter or packet delay variation, that can be calculated in real-time as the floating average of differences between the timestamps, contained in the RTP protocol header, of consecutively received packets [17][18].

Jitter can affect heavily the perceived quality of a voice call, and for this reason, at the receiver side, de-jitter buffers are implemented, with the aim of re-establishing the right order of IP packets, by adding to each packet a proper delay.

If the end-to-end delay of a packet is greater than de-jitter buffer dimension, the packet is discarded. The consequence of this operation is a degradation of voice quality [19][20].

Concerning VoLTE QoE, the most relevant KQI is the Mean Opinion Score (MOS), an adimensional subjective parameter for the evaluation of voice call quality, with values in the range between 1 and 5 [19][20].

MOS can be estimated by the E-Model algorithm, whose output is the R-Factor, defined in [21]-[24]. Table I below illustrates the matching between R-factor and MOS values.

The R-factor can be expressed as follows, according to [21]:

$$R = R_0 - I_s - I_d - I_{e,eff} + A. \quad (1)$$

where:
- R_0 is the Signal to Interference Ratio;
- I_s is a combination of impairment factors, occurring simultaneously or not in the voice session;
- I_d is an impairment factor due to talk and listener echo, and delay contributions;
- A is known as Expectation or Advantage Factor, with values from 0 to 20, as shown in table 2/G.107 in [18];
- $I_{e,eff}$ stands for Equipment Factor, and represents the impairment caused by low bit rate CODEC, and packet loss. In the ITU-T G.107 recommendation [21], this parameter has been expressed by:

$$I_{e,eff} = I_e + (95 - I_e)\frac{P_{pl}}{P_{pl}+B_{pl}}. \quad (2)$$

where I_e and B_{pl} are parameters that can assume values as indicated in the recommendation, and P_{pl} is the packet loss, in a range of values between 0 and 1, calculated as:

$$P_{pl} = 1 - \frac{m}{n}. \quad (3)$$

where m is the number of RTP packets received and n is the number of RTP packets sent, with uncorrelated losses [21]. Packet loss is an important KPI to take into account, as seen further on.

III. RELATED WORKS

The analysis of User Experience in real time services like Voice over IP (VoIP), VoLTE and Video over LTE (ViLTE) has been the object of several studies in literature, taking into account voice services delivered over an IP core network, and a radio access network that can be LTE access network or Wi-Fi and WiMAX one.

In [25] the impact of QoS parameters (such as packet loss and packet delay) on the QoE of ViLTE is evaluated by using a test-bed integrating the real network with an UE under test.

TABLE I. R-FACTOR MOS MATCHING [18]

User Satisfaction Level	R-Factor	MOS
Maximum using G.711	93	4.4
Very satisfied	90-100	4.3-5
Satisfied	80-90	4-4.3
Some users satisfied	70-80	3.6-4
Many users dissatisfied	60-70	3.1-3.6
Nearly all users dissatisfied	50-60	2.6-3.1
Not recommended	Less than 50	1-2.6

In [26], the MOS is used to adaptively control the QoE of VoIP services periodically determined by means of a modified version of E-Model.

In [27], by using an OPNET simulation model, MOS, delay and jitter have been observed in a VoIP application, with a network scenario in which the radio access system is either Wi-Fi or WiMAX.

In [28], simulation results are presented that show the improvements that can be obtained in the VoLTE MOS by use of a closed-loop power control algorithm applied to the downlink of the VoLTE radio bearer for an indoor scenario served by small cells.

In [29], the MOS of VoIP services in 3G mobile networks is evaluated in applications like Line and Skype. QoS parameters as jitter and packet loss were measured and the QoE was derived using the QoS versus QoE tables defined by the G.107 ITU-T [21].

In [30], a dynamic adaptation algorithm of joint source-channel code rate is used to improve the VoLTE QoE. The wideband E-Model is used to assess the voice quality.

All mentioned works perform QoE measurements on actual platforms.

Unfortunately, QoE measurements are to be performed on the user devices (rather than on the network), but the measurements of this type need expensive user test equipment.

For this reason, this paper approach is to mathematically derive QoE from QoS, since QoS measurements can be easily obtained directly on the network, by use of standard equipment.

In literature, a series of studies can be found that derive the QoE from QoS by use of a mathematical relationship holding between the two [31]-[33].
However, such studies refer to VoIP services, while in this paper the mathematical relationship holding for VoLTE is derived.

IV. THE VOLTE QOE MODEL

A mathematical relationship between QoS and QoE holding for VoIP has been introduced in [32][33]. This relationship is known as the *IQX Hypothesis (exponential interdependency of Quality of Experience and Quality of Service)*.

In such a relationship, QoE depends on only one QoS parameter, and it is expressed as follows:

$$QoE = \alpha e^{-\beta QoS} + \gamma. \quad (4)$$

where α, β and γ in (4) are calculated by means of non-linear regression. An example of regression result is as follows:

$$f(p_L) = 2.861e^{-29.816 p_L} + 1.134. \quad (5)$$

where the QoE, in MOS, is indicated with $f(p_L)$ and the QoS parameter is packet loss [32][33].

The IQX hypothesis has been drawn up for an IP context, and it is related to a particular service, the voice call, so it can represent a good starting point in establishing QoE/QoS relationship for VoLTE.

The IQX hypothesis deals with voice application and with a QoE/QoS relationship based on only one QoS parameter (either jitter or packet loss). In this paper, instead, the QoE of voice application is assessed by basing the relationship on two QoS parameters (both jitter and packet loss).

To this scope, in the model presented here, the strict packet loss expression (3) will be replaced by an expression that includes the contribution of packet loss and jitter, known as effective packet loss $P_{pl,eff}$, defined in [31]:

$$P_{pl,eff} = 1 - \left(1 - P_{pl}\right)\left(1 - P_{jitter}\right). \quad (6)$$

where P_{pl} remains the packet loss from (3), and jitter is expressed as a Pareto probability by writing:

$$P_{jitter} = \frac{1}{2}\left(1 - \frac{0.1x}{\sigma}\right)^{20}. \quad (7)$$

where x is the jitter buffer dimension and σ is network jitter delay, both expressed in [ms].

The $P_{pl,eff}$ in (6) can be considered as a further KPI resulting from the combination of two original KPIs, network jitter and packet loss [20].

By replacing (6) in the exponent of (4), the expression for MOS becomes:

$$MOS = \alpha e^{-\beta P_{pl,eff}} + \gamma. \quad (8)$$

Apart from its academic interest, the equation (8) could be useful for the network operator to predict the User Experience starting from the variation of the combined jitter and packet-loss parameters. The next Section provides the necessary validation of (8) in the VoLTE context.

V. VALIDATION OF THE QOE VERSUS QOS RELATIONSHIP FOR VOLTE SERVICES

The use of simulation is the first step for the comprehension of network mechanisms that can affect performances and services. A well done simulation model can help to get the perspective view of the network, without turning to expensive ad hoc experimental solutions, used in real scenarios.

In this research, OPNET Modeler [34] has been used to simulate an LTE network with the IMS section, since OPNET offers a plurality of modules and network nodes compliant with the 3GPP LTE standard. Moreover, OPNET also allows to monitor the network performance by placing probing points on the simulated network.

An OPNET simulation model gives in output KPIs like jitter and packet loss and system KQIs like MOS. These parameters are evaluated at application level.

Simulation parameters such as number of simulation runs or initial bias removal are taken care of internally by OPNET, in order to guarantee the statistical significance of the results.

Figure 1 gives the geographical overview of the simulated scenario. The IMS, (see Sect. II), using Session Initiation Protocol (SIP) application signaling protocol, is responsible for initiating, maintaining and terminating VoLTE call set-up in LTE.

In the model, IMS is physically located in Milan and it is composed of standardized nodes Proxy Call Session Control Function (P-CSCF), Serving Call Session Control Function (S-CSCF) and Interrogating Call Session Control Function (I-CSCF), simulated by means of proxy servers, linked to Gtwy2 by a 1000BaseX link [14][15].

The Gtwy2 and IP_Backbone are linked via the PPP_DS1, and the IP_Backbone is linked to Gtwy1 by a similar link. Gtwy1 and Gtwy2 are simulated by an OPNET Ethernet4_slip8_gtwy, i.e., a router with 4 Ethernet and 8 IP interfaces.

A "Campus Network" node, physically located in Rome, and detailed in Figure 2, is linked to Gtwy1 via a 1000BaseX link. The Campus Network is a LTE network, with physical dimension of 10x10 Km, composed of various OPNET elements:

- An Evolved Packet Core (EPC) node, fully compliant with the standard;
- Two eNodeB nodes.
- Eight mobile workstations (UE_1_1, UE_1_2, UE_1_3_t and UE_1_3_t2 attached to eNB_1; and UE_2_1, UE_2_2, UE_2_3_t, UE_2_3_t2 attached to eNB_2), to represent eight User equipment generating and receiving IP traffic.
- A File Transfer Protocol (FTP) server.
- An Application Definition Node, for the characterization of application parameters.
- A Profile Definition Node, creating the profiles that generate the traffic in a specific temporal order, by means of the applications defined with the Application Definition Node.
- An LTE Configuration Node, setting physical LTE configurations and EPS Bearer specification.
- An IP QoS Node, for the definition of scheduling policies.

The connections between the two eNodeBs and the EPC, and between the EPC and the FTP server simulate a 1000BaseX link (44.736 Mbps).

The propagation settings of the radio interface section between UEs and eNodeBs simulate a typical urban scenario.

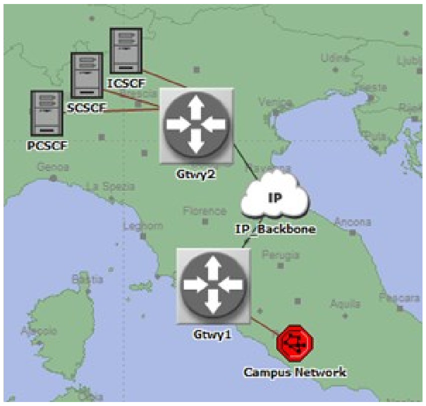

Figure 1. Geographical overview of the simulated scenario.

VI. THE SIMULATED TRAFFIC LOAD

In order to simulate a realistic network scenario, in addition to VoLTE traffic, background traffic has also been generated. It consists of FTP, Voice over IP and Video Streaming applications.

The application protocols taken into account are RTP for VoLTE and VoIP over LTE, and Real Time Streaming Protocol (RTSP) for Video Streaming.

For what concerns FTP, it consists of a file transfer from a server to the UE, so the transport protocol is User Datagram Protocol (UDP).

The characteristics of monitored and background applications are:

- VoLTE (monitored application)
  Codec: RTP Adaptive Multi Rate (AMR) 12.2K
  Frame size: 10 ms
  Code rate: 64 kbps
  Uplink Guaranteed Bit Rate: 1 Mbps
  Downlink Guaranteed Bit Rate: 1 Mbps
  ToS: Interactive Voice - 6
  QoS Class Identifier (QCI): 1
  Allocation and Retention Priority (ARP) = 1
- FTP application (background traffic)
  Packet dimension: 1000 byte
  FTP get, UE download file from FTP server
  Type of Service (ToS): Best Effort - 0
  ARP = 5
- Voice application (background traffic)
  Codec: RTP AMR 12.2K
  Frame size: 10 ms
  Code rate: 64 kbps
  ToS: Interactive Multimedia - 5
  ARP = 2
- Video Streaming application (background traffic)
  ToS: Interactive Multimedia - 5
  ARP = 4

The simulated run time was for 800 sec. The VoLTE application starts 170 sec after the simulation begins, and it remains active until simulation ends.

The FTP application starts 135 sec after the simulation begins, and it is composed of 2 active blocks of 120 sec each, with an inter-repetition time of 300 sec between the blocks.

The Voice and Video Streaming applications both start 170 sec after the simulation begins, and they remain active until the simulation ends.

The VoLTE application runs between UE_1_1 (the caller) and UE_2_1 (the called).

The FTP application runs between UE_1_2, UE_2_2, and the FTP server.

The Voice application runs between UE_1_3_t (the caller) and UE_2_3_t (the called).

The Video Streaming application runs between UE_1_3_t2 and UE_2_3_t2.

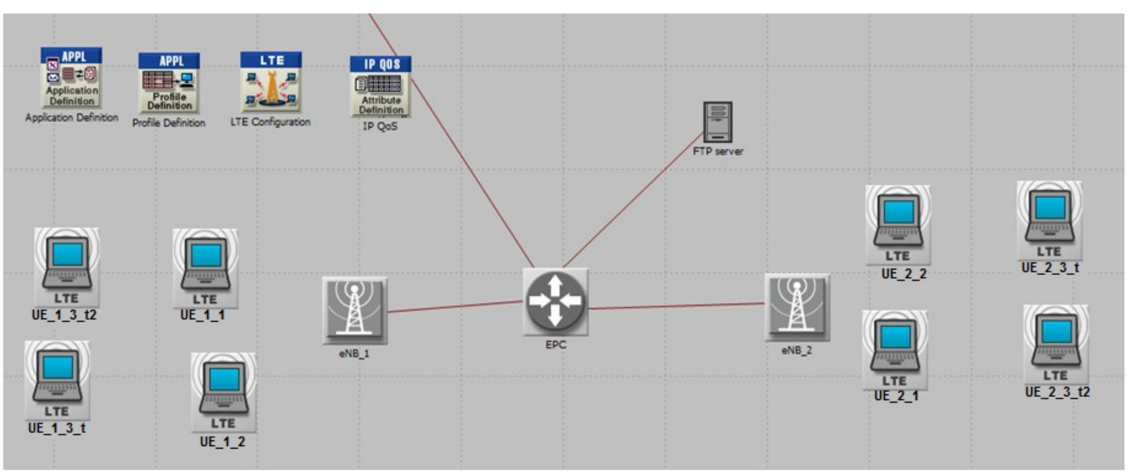

Figure 2. Campus network in detail

VII. ANALYSIS OF THE SIMULATION OUTPUT

OPNET has been used to investigate the trend of the QoE (MOS) versus increasing values of jitter and packet loss. To obtain variations in the VoLTE jitter and packet loss, various changes in the above described four traffic loads have been operated.

By setting the timing of such traffic loads, the VoLTE traffic has been monitored, and performance parameters have been collected.

Several scenarios have been simulated versus increasing values of jitter and packet loss and simulation results have been extracted by means of probes placed on the network nodes and interfaces.

The values of the network KPIs (VoLTE network jitter and VoLTE packet loss) and the KQI (VoLTE MOS) obtained from simulation have been analyzed with the Matlab regression toolbox in order to estimate the α, β and γ parameters holding for VoLTE.

In a first approach, simulation has been used to check under which condition the IQX Hypothesis in (4), originally introduced for VoIP, could also hold for VoLTE. To this purpose, the behavior of the VoLTE MOS was plot versus the network jitter first, and then versus the packet loss P_{pl} defined in (3).

Figure 3 gives the plot of MOS versus network jitter and shows that its mathematical regression (with a coefficient of correlation 0.998) well complies with the exponential behavior of (4). This proves that relationship (4), originally introduced for VoIP, also holds for VoLTE services.

Figure 4 instead shows that no exponential relationship holds between the VoLTE MOS and packet loss. In other words, the IQX Hypothesis can be extended to VoLTE, under condition that the QoS parameter appearing in the exponent of (4) is network jitter, while no extension holds in case the exponent is packet loss P_{pl}.

The question then arises: what happens if, in place of the P_{pl} defined in (3), the $P_{pl,eff}$ defined in (6) is instead used in the exponent, i.e. if a combination of jitter and packet loss is used? In other words, what happens if the VoLTE MOS is expressed by (8)?

Figure 5 shows that the mathematical regression of the simulation MOS versus $P_{pl,eff}$ well complies with the exponential behavior of (8) and this gives the expected validation of formula (8) for the VoLTE context. To strengthen such a proof, the VoLTE MOS produced from simulation was plot versus P_{jitter} defined in (7) and versus the standard packet loss P_{pl} defined in (3), taken separately.

The result can be seen in Figure 6 where the simulation output is compared with the mathematical 3D regression obtained using the two QoS parameters separately. The exponential relationship is shown to hold.

This completes the simulation validation of model (8). Such validation has been confirmed by real world tests performed on the VoLTE national scenario, in cooperation with the TIM telecommunication company, on the occasion of a patent filing [35].

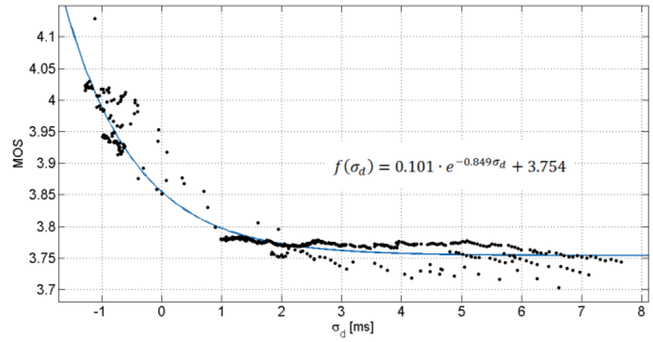

Figure 3. Behavior of VoLTE MOS versus network jitter.

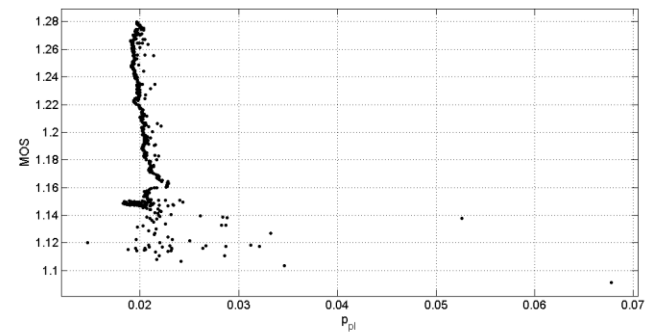

Figure 4. Behavior of VoLTE MOS versus packet loss P_{pl}.

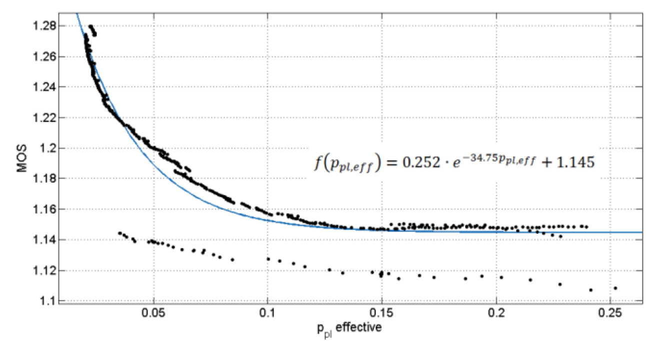

Figure 5. Behavior of VoLTE MOS versus effective packet loss $P_{pl,eff}$.

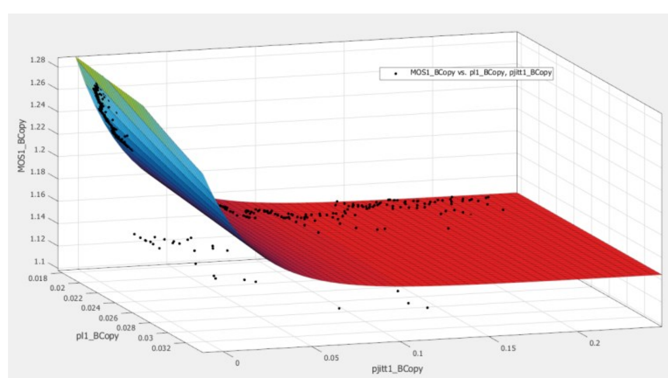

Figure 6. Behavior of VoLTE MOS versus packet loss P_{pl} and P_{jitter}.

VIII. CONCLUSIONS

The LTE standard only focuses on the definition of network QoS, without giving its relationship with the QoE. In this paper, a hypothesis relationship was introduced and proved valid for VoLTE services by use of an OPNET simulation model. Validation was confirmed by real world tests performed in cooperation with the telecommunication company TIM, on the VoLTE national scenario.

Knowing this relationship, the network management can adjust the network KPIs (packet loss and jitter) to continuously meet the QoE expected from the SLA negotiated with the end-user.

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