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A New Non-intrusive Model for Measuring VoLTE Quality based on Wideband E-model

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ABSTRACT

Long Term Evolution (LTE) was initially designed for a high data rates network. However, voice service is always a main service that brings a big benefit for mobile operators. Hence the deployment of Voice over LTE (VoLTE) is very essential and urgent. LTE network is a fully All-IP network, thus, the deployment of VoLTE is very complex, specially for guaranteeing of Quality of Service (QoS) to meet quality of experience of mobile users. The key purpose of this paper is to present an object, non-intrusive prediction model for VoLTE quality based on LTE-Sim framework and Wideband (WB) E-model. In order to simulate the experiment, we deployed LTE-Sim framework combining C programming in Matlab software. The outputs of the LTE-Sim will be the inputs of the WB E-model. The output of the WB E-model is user satisfaction via Mean Opinion Score (MOS). In addition, due to voice over LTE network (VoLTE) uses Adaptive Multi-Rate Wideband (AMR-WB) for vocoder while LTE-Sim supports only G.729 codec for narrow band, thus, we propose to add AMR-WB codec into LTE-Sim software. The simulation results show that the proposed model can predict voice quality in LTE network via the WB E-model. The proposed model does not refer to the original signal, thus, it is very suitable for predicting voice quality in LTE network for many different scenarios which are configured in the LTE-Sim framework. The proposed model is very suitable for predicting VoLTE quality for transmission planning, and for researchers in Laboratory.

Keywords

VoLTE, voice quality, LTE, Wideband E-model, MOS

1. INTRODUCTION

LTE network is developed by the Third Generation Partnership Project (3GPP) [3]. It is a mobile network which has high data rates, low latency and is fully packet-based. This means to improve the capability of legacy system by increasing data rates and extending superior QoS for various multimedia applications. Voice over LTE network (called VoLTE) is a main service of LTE network. Since LTE network is a full packet-switched, thus, the deployment of VoLTE service is very complicated. All voice traffics over LTE network are VoIP (Voice over Internet Protocol) including VoLTE. According to [12], there are two types of voice traffic over LTE network, those are VoLTE and VoIP. VoLTE is really the VoIP service with QoS guaranteed [12]. Table 1 represents the service classes in LTE network where voice service is a Guaranteed Bit Rate (GRB) service which has the second priority just after IP Multimedia Subsystem (IMS) signaling. However, in order to guarantee VoLTE quality is an extreme challenge.

In communications systems, the perceived voice quality is usually represented as the MOS. MOS can be attained by many methods. These methods are divided into two groups called subjective methods and objective ones. Subjective methods humans listening to a live stream or a recorded file and rating it on a ratio of 1 (poor) to 5 (excellent) [13]. These methods have some disadvantages such as too expensive, time consuming and are not suitable for a large network infrastructure. Otherwise, objective methods have more advantages, they eliminate the limitations of subjective methods. Objective methods are classified into two approaches: intrusive and non-intrusive ones. The intrusive methods (e.g. Perceptual evaluation of speech quality for Wideband audio (PESQ-WB) [14]) are more exact and are widely utilized to predict aware voice quality. However, they are not suitable for real-time services such as VoIP because they require original signals to refer. The non-intrusive methods (e.g. ITU-T WB E-model [15]) are computational models that are used for transmission planning purposes. They are not as accurate as the intrusive approaches and they do not have complex mathematical operations. The obtained results from objective methods do not always well relate to human perception. The main advantage of the non-intrusive methods are they predict voice quality without any reference to the original signals and they require less parameters than the intrusive methods. For several typical non-intrusive methods, authors in [23] proposed to use Random Neural Network (RNN) to assess voice quality over internet. Voice quality assessment was predicted in [16] using RNN. Another non-intrusive method was proposed in [11] based on RNN for evaluating video quality in LTE network. Authors in [10] proposed to use WB E-model to predict VoLTE quality for minimizing redundant bits generated by channel coding. In [9], authors investigated the effects of Packet Loss Rate (PLR) and delay jitter on VoIP quality to assess prediction errors of MOS for the E-model. A new model called “Packet-E-Model” was proposed in [18] to measure speech quality perception for VoIP in Wimax network. In [6], authors presented a voice quality measurement tool based on the E-model. A framework of objective assessment method for estimating conversational quality in VoIP was proposed in [28]. In [5], a simplified versions of the E-model were proposed to simplify the calculations and focus on the most important factors required for monitoring the call quality. According to our knowledge, at present, there are not any

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Table 1. LTE service classes with QoS requirements

Resource Type	Priority	Packet Delay Budget (ms)	Packet Error Loss Rate	Example services
Guaranteed Bit Rate (GBR)	2	100	10^{-2}	Conversational voice
	4	150	10^{-3}	Conversational video (live streaming)
	3	50	10^{-3}	Real-time gaming
	5	300	10^{-6}	Non-conversational video (buffered stream)
Non-GBR	1	100	10^{-3}	IMS signaling
	6	300	10^{-6}	Video (buffered streaming) TCP-based (e.g. www, e-mail, chat, FTP, P2P sharing, progressive video, etc.)
	7	100	10^{-6}	Voice, Video (live streaming, Interactive Gaming)
	8	300	10^{-3}	Video (buffered streaming) TCP-based (e.g. www, e-mail, chat, FTP, P2P sharing, progressive video, etc.)
	9		10^{-6}	

proposals which allow to predict VoLTE quality using LTE-Sim framework and WB E-model.
We see that, the lack of the WB E-model is how to determine exactly the its input parameters. Therefore, we need a method to measure the input factors of the WB E-model. In this paper, we propose to use the LTE-Sim [21] to calculate delay and PLR for voice users. LTE-Sim is a famous framework which allows to simulate entire LTE network, thus, it is quite similar to a real system. Outputs of LTE-Sim are Delay and PLR which are essential input parameters of the WB E-model. In addition, we proposed to complement the AMR-WB codec into LTE-Sim software to overcome a limitation of the LTE-Sim is that it supports only G.29 codec for VoIP application. In order to obtain more real results, we simulate voice service with mobility in LTE heterogeneous network.
The remainder of this paper is organized as follows: Overview of the system model is described in section 2. In section 3, we present the proposed model. The simulation results and performance evaluation of the proposed model are analysed in section 4. The conclusion and future work is represented in section 5.

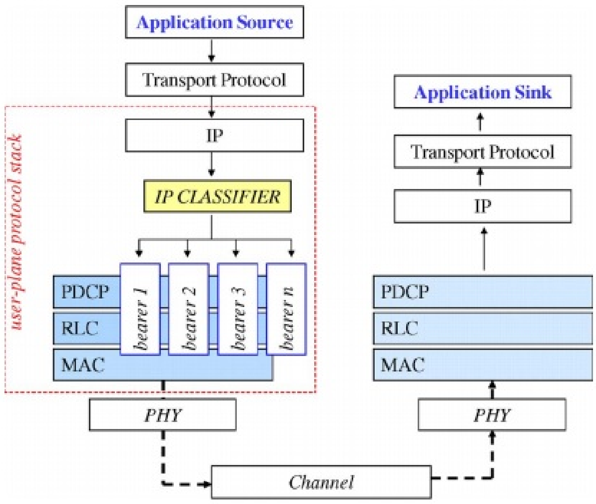

Fig. 1. The implemented protocol stack in LTE-Sim [21]

2. THE SYSTEM MODEL

2.1 The LTE-Sim framework

LTE-Sim is an open-source framework which is developed by Giuseppe Piro and his colleagues [21]. It is freely available for scientific community. It is used to simulate entire LTE network. There are many researchers who used LTE-Sim to simulate their proposals such as scheduling strategies, radio resource optimization, frequency reuse techniques, the adaptive modulation and coding module, user mobility, and etc. for both downlink and uplink directions and in both multicell/multiuser environments. The implemented protocol stack of LTE-Sim for user-plane is represented on Figure 1. It is nearly similar to a real LTE system.

Figure 1 can be briefly described as follows: When a voice traffic flow transmitted over the LTE-Sim, it is encapsulated sequentially with network protocols. For the downlink direction, the VoLTE packet uses transport protocols of Real-time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP). It is then packetized with radio protocols such as Packet Data Convergence Protocol (PDCP), Radio Link Control (RLC) and Medium Access Control (MAC), and Physical (PHY) layer before it is transmitted over the air interface. LTE-Sim supports both IPv4 and IPv6 protocols with header sizes are 40 and 60 bytes, respectively while the voice payload is about 32 bytes, thus, to reduce the overhead, Robust Header Compression (RoHC) is deployed at PDCP layer. The IP header is then compressed by RoHC down to only 1-4 bytes, normally 3 bytes.

LTE-Sim allows to simulate a heterogeneous traffic over LTE network. Therefore, it is quite similar to a real system. The outputs of this software including many parameters where there are delay and PLR. LTE-Sim supports multicell/multiuser with mobility in a heterogeneous network. The details of the simulation scenario is represented in Section 3.

2.2 VoLTE source codec

VoLTE uses AMR-WB as a vocoder. AMR-WB codec is a speech codec which has been developed by ETSI (the European Telecommunications Standards Institute) and applied in the 3GPP LTE network for voice compression and decompression. It is fully described in [24]. AMR-WB codec uses a sampling rate of 16 kHz, which covers 50-7000 Hz audio bandwidth. It has 9 different codec modes (from mode 0 to mode 8) corresponding to 9 source bit rates in range of 6.6-23.85 Kb/s. Each of them generates an encoded 20 ms speech frame and switches among them every 20 ms. The bits in the encoded speech frame are ordered according to their subjective importance. These bits are divided into three classes with reducing perceptual importance: Class A, Class B and Class C. Total bits of each class depend on codec mode. AMR-WB packet size depends on the bit rate (mode) is described such as in Table 2 [8].
In LTE network, AMR-WB codec is configured into 3 configurations [1] as follows:

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Table 2. Packet sizes of AMR-WB codec modes

Parameter	AMR-WB bit rate (kbps)								
	23.85	**23.05**	**19.85**	**18.25**	**15.85**	**14.25**	**12.65**	**8.85**	**6.6**
Payload size (bits)	477	461	397	365	317	285	253	177	132
Frame size (bits)	488	472	408	376	328	296	264	192	144
RTP header (bits)	96	96	96	96	96	96	96	96	96
Packet size (bits)	**584**	**568**	**504**	**472**	**424**	**392**	**360**	**288**	**240**

—Configuration A (Config-WB-Code 0): 6.6, 8.85, and 12.65 Kb/s (Mandatory multi-rate configuration).
—Configuration B (Config-WB-Code 2): 6.6, 8.85, 12.65, and 15.85 Kb/s.
—Configuration C (Config-WB-Code 4): 6.6, 8.85, 12.65, and 23.85 Kb/s.

These configurations are used to simplify the negotiation of bit rate between the user equipment and the base station, thus will simplify the implementation and testing. The remaining bit rates can still be used for other purposes in mobile networks. In order to choose a bit rate, the receiver measures quality of radio channel. The channel quality indicator (CQI) is used for this purpose. It is defined as an equivalent carrier-to-interference (C/I) ratio. The C/I ratio then compared to a set of predefined thresholds to decide which mode to be used. Switching among modes in a configuration depend on the rate control algorithm in AMR-WB codec. The criterion for mode switching is threshold value of C/I ratio. These threshold values depend on the channel condition, frequency hopping scheme, network configuration and other factors. Furthermore, network conditions change over time, so that, even well-selected adaption thresholds will not be best.

2.3 WB E-model: Speech quality assessment for Wideband Audio

WB E-model is a computational model developed and standardized by ITU-T [25]. It is used to estimate the MOS for wideband audio quality. The output of the model is R-factor. The values of this R-factor in range of 0-129. And then, it is mapped to the MOS. The R-factor in the WB E-model is defined as follows:

$$R_{wb} = R_{0,wb} - I_{s,wb} - I_{d,wb} - I_{e,eff,wb} + A \quad (1)$$

In which:

—$R_{0,wb}$: The basic signal-to-noise ratio;
—$I_{s,wb}$: The simultaneous impairment factor, it is the sum of all impairments which may occur more or less simultaneously with the voice transmission. In this model, this factor is set to 0;
—$I_{d,wb}$: The delay impairment factor, representing all impairments due to delay of voice signals;
—$I_{e,eff,wb}$: The equipment impairment factor, capturing the effect of signal distortion due to low bit rates of the codec and packet losses of random distribution;
—A: The advantage factor, capturing the fact that some users can accept a reduction of quality due to the mobility of cellular networks. In this model, this factor is set to 0.

In above factors, $I_{d,wb}$ and $I_{e,eff,wb}$ are affected by end-to-end delay and packet loss, respectively, while $R_{0,wb}$ and $I_{s,wb}$ do not depend on network performance. The R_{wb} factor is translated into the MOS as follows [25]:
For $R = R_{wb}/1.29$

$$MOS = \begin{cases} 1, \textbf{if } R < 0 \\ 1 + 0.035 \times R + 7 \times 10^{-6} \times R \times \\ (R-60) \times (100-R), \textbf{if } 0 \leq R \leq 100 \\ 4.5, \textbf{otherwise} \end{cases} \quad (2)$$

The relation between R-factor, user perception, and MOS is described in the Table 3.

Table 3. R-factor and MOS with corresponding user satisfaction

R	**User satisfaction**	**MOS**
$90 \leq R < 100$	Very satisfied	4.3-5.0
$80 \leq R < 90$	Satisfied	4.0-4.3
$70 \leq R < 80$	Some users dissatisfied	3.6-4.0
$60 \leq R < 70$	Many users dissatisfied	3.1-3.6
$50 \leq R < 60$	Nearly all users dissatisfied	2.6-3.1
$R < 50$	Not recommended	< 2.6

R_{wb} factor is then mapped to the MOS using Equation (2), and then, the MOS is mapped to the satisfaction level of the users. According to [19], for the wideband audio, the value of $R_{0,wb}$ factor in equation (1) equals 129, thus, equation (1) can be rewritten as follows:

$$R_{wb} = 129 - I_{d,wb} - I_{e,eff,wb} \quad (3)$$

In order to compute the R_{wb} factor, we have to count the values of $I_{d,wb}$ and $I_{e,eff,wb}$ factors. The $I_{d,wb}$ factor is determined by the following equation [20]:

$$I_{d,wb} = 0.024 \times D_{e2e} + 0.11 \times (D_{e2e} - 177.3) \times H(D_{e2e} - 177.3) \quad (4)$$

In which: $H(x)$ is the Heavyside function:

$$H(x) = \begin{cases} 0, \textbf{if } x < 0 \\ 1, \textbf{otherwise} \end{cases} \quad (5)$$

In equation (4), D_{e2e} represents the total end-to-end delay of speech packet. It can be obtained after finishing the simulation scenario in the LTE-Sim software. The $I_{e,eff,wb}$ is determined according to packet loss. According to [19], $I_{e,eff,wb}$ is determined as follows:

$$I_{e,eff,wb} = I_{e,wb} + (129 - I_{e,wb}) \times \frac{P_{pl}}{P_{pl}+B_{pl}} \quad (6)$$

Where: $I_{e,wb}$: The respective impairment factor without any packet loss. P_{pl}: Packet loss rate. It can be also obtained after finishing the simulation scenario in the LTE-Sim software. B_{pl}: A codec-specific factor which characterizes its robustness against packet loss. The values of $I_{e,wb}$, B_{pl} is represented in Table 4 [19].

Table 4. Values of $I_{e,wb}$ and B_{pl} according to AMR-WB modes

AMR-WB mode	**Bitrate (bps)**	$I_{e,wb}$	B_{pl}
0	6.6	39	12.8
1	8.85	25	13.5
2	12.65	11	13
3	14.25	10	14.1
4	15.85	7	13.1
5	18.25	5	12.5
6	19.85	4	12.3
7	23.05	1	13
8	23.85	6	12.2

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The R-factor is then mapped to the MOS via equations (2). MOS score performs the user perception.

3. THE PROPOSED MODEL

Voice over LTE network (VoLTE) is a real-time service, and is fully deployed over an IP network, thus, guarantee of VoLTE quality is a big challenge. There are very little methods that allow to monitor and to predict VoLTE quality and most of them are subjective ones which have to refer to the original signal, thus, they are not suitable for real-time services such as VoLTE. In this paper, we propose a new non-intrusive voice quality assessment method which based on the LTE-Sim software [21] and the Wideband E-model [25]. LTE-Sim is a software which allow to simulate VoIP flow that is quite similar to a real flow. Therefore, we see that, the simulation results are quite exact. The WB E-model is a computational model which allows to predict voice quality when it is transmitted from source to destination. Such as mentioned previously, VoLTE uses AMR-WB as a vocoder and this is mandatory. However, LTE-Sim supports only G.729 codec which has only a bitrate of 8 kbps and generates a packet of 32 bytes in each 20 ms. Each birate of AMR-WB corresponds to a mode. AMR-WB modes are configured into 3 configurations such as described above. Its mode is adapted according to the channel quality. Therefore, in order to add this codec into LTE-Sim, we have to build a procedure which allow to change instantly the AMR-WB mode according to the channel quality via Signal-to-interference (C/I) ratio. The C/I ratio is calculated at the receiver and is feedbacked to the eNodeB. The changes among the AMR-WB modes depends on C/I thresholds. The proposed procedure for changing the AMR-WB modes (or changing packet size of AMR-WB codec) for the configuration A, B, and C are represented in Algorithm 1, Algorithm 2, and Algorithm 3, respectively. The values of C/I thresholds are referred to [29] and [30]. Besides, we also have several essential modifications in LTE-Sim to suit for the simulation scenario.

Algorithm 1 Change AMR-WB packet size according to the channel quality for the **Config-WB-Code 0**

procedure UPDATE AMR-WB PACKET SIZE
Step 1: Calculate C/I ratio: available in LTE-Sim
Step 2: Update packet size of AMR-WB codec
  if calculated $C/I \leq 6.5$ **then**
    $mode = 0$
    Update $Packet\_size = 30$
  else if calculated $C/I \leq 12.5$ **then**
    $mode = 1$
    Update $Packet\_size = 36$
  else
    $mode = 2$
    Update $Packet\_size = 45$

The proposed model is represented on Figure 2. The principle

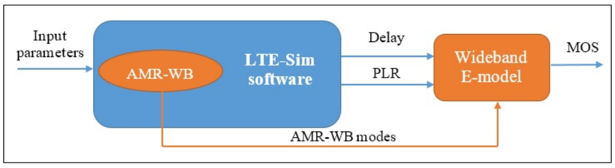

Fig. 2. The implemented model

of the proposed model as follows: Input parameters of the simulation scenarios are firstly fed to the LTE-Sim software. After finishing the simulation, we receive simulation results. In these

Algorithm 2 Change AMR-WB packet size according to the channel quality for the **Config-WB-Code 2**

procedure UPDATE AMR-WB PACKET SIZE
Step 1: Calculate C/I ratio: available in LTE-Sim
Step 2: Update packet size of AMR-WB codec
  if calculated $C/I \leq 6.5$ **then**
    $mode = 0$
    Update $Packet\_size = 30$
  else if calculated $C/I \leq 12.5$ **then**
    $mode = 1$
    Update $Packet\_size = 36$
  else if calculated $C/I \leq 18.5$ **then**
    $mode = 2$
    Update $Packet\_size = 45$
  else
    $mode = 4$
    Update $Packet\_size = 53$

Algorithm 3 Change AMR-WB packet size according to the channel quality for the **Config-WB-Code 4**

procedure UPDATE AMR-WB PACKET SIZE
Step 1: Calculate C/I ratio: available in LTE-Sim
Step 2: Update packet size of AMR-WB codec
  if calculated $C/I \leq 6.5$ **then**
    $mode = 0$
    Update $Packet\_size = 30$
  else if calculated $C/I \leq 12.5$ **then**
    $mode = 1$
    Update $Packet\_size = 36$
  else if calculated $C/I \leq 18.5$ **then**
    $mode = 2$
    Update $Packet\_size = 45$
  else
    $mode = 8$
    Update $Packet\_size = 73$

results, we select Delay, PLR according to the number of user. These factors are the input parameters of the WB E-model. The output of WB E-model is R-factor, and then it is mapped to MOS score. This MOS score performs the user satisfaction. The steps of the proposed model can be described as follows:

—**Step 1:** Setting input parameters of the simulation scenario.
—**Step 2:** Complement the procedure of changing AMR-WB packet size according to the C/I ratio into LTE-Sim.
—**Step 3:** Simulating the simulation scenario in the LTE-Sim software.
—**Step 4:** Collecting the delay, PLR results from Step 3, and selecting values of AMR-WB modes.
—**Step 5:** Feeding the input parameters taken from Step 3 to the WB E-model which is programmed in Matlab software [17] as follows:
  —Calculating R-factor of the WB E-model according to Equation (3).
  —R is then mapped to MOS based on Formula (2).
  After Step 5, we will receive the user satisfactions according to the number of user.

LTE is a packet-switched network, thus, in order to simulate it as a real system, we select input data flows as follows: Each user utilizes a VoIP, a Video and a non real-time flow at an instant. This means the simulation scenario is executed in a heterogeneous LTE network. In addition, the mobility is also included in the simulation scenario, specifically, we select the speed of user is 30 km/h. We choose the Frame Level Scheduler (FLS) [22], Modified Largest Weighted Delay First (M-LWDF) [4], and Exponential/Proportional Fair (EXP/PF) [27] schedulers which are

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very suitable for real-time services. Besides predicting the user perception, we assess the effects of the Delay, PLR on user satisfaction according to the number of user.

4. SIMULATION ENVIRONMENT AND PERFORMANCE EVALUATION

4.1 Simulation environment

4.1.1 Traffic model. In the simulation scenario, the eNB is located at the center of the macrocell using an ommi-directional antenna in a 10 MHz bandwidth. Each UE uses a VoIP flow, a Video flow, and a INF-BUF flow at the same time. For the VoIP flow, a G.729 voice stream with a bit-rate of 8 kbps was considered. The voice flow is a bursty application that is modelled with an ON/OFF Markov chain [7]. For the video flow, a trace-based application that generates packets based on realistic video trace files with a bit-rate of 242 kbps was used in [26] and it is also available in [21]. In order to obtain a realistic simulation of an H.264 SVC video streaming, we used an encoded video sequence "*foreman.yuv*", which is publicly available. The LTE propagation loss model is formed by four different models including: Path loss, Multipath, Penetration and Shadowing [2].

—Path loss: $PL = 128.1 + 37.6 \times log(d)$, with d is the distance between the UE and the eNB in km.
—Multipath: Jakes model
—Penetration loss: 10 dB
—Shadowing: Log-normal distribution with mean 0 dB and standard deviation of 8 dB.

4.1.2 Simulation parameters. The simulation process is performed in a single cell with interference with the number of users in the interval [10, 50] which move randomly at a speed of 30 km/h. The other basic parameters used in the simulation are represented in the Table 5.

Table 5. Simulation parameters

Simulation Parameters	Values
Simulation duration	100 s
Frame structure	FDD
Cell radius	1 km
Bandwidth	10 MHz
Video bit-rate	242 kbps
VoIP bit-rate	6.6, 8.85, 12.65 kbps
User speed	30 km/h
Number of user	10, 20, 30, 40, 50 UEs
Maximum delay	0.1 s
Packet Scheduler	FLS, M-LWDF, EXP/PF
Traffic model	VoIP, Video, and INF-BUF

4.2 Performance evaluation

The analyses of the simulation results are represented in the following subsections.

4.2.1 Effects of AMR-WB modes on voice quality. Figure 3 represents the effects of AMR-WB mode 0 on voice quality. Is is clear that, the voice quality slightly decreases when the number of VoIP user (NU) increases excepting the FLS scheduler. The FLS scheduler has the best quality compared to the other schedulers, even, it does not decrease when the NU raises. The M-LWDF scheduler has the worst quality and it seems to heavily decreases when the NU increases.
For the effects of AMR-WB mode 1 on user perception such as illustrated on Figure 4, it is easy to see that user satisfaction increases when the mode of AMR-WB increases. This is suitable. It is similar to the case of AMR-WB mode 0, the FLS scheduler still keeps the first position of highest MOS while the M-LWDF

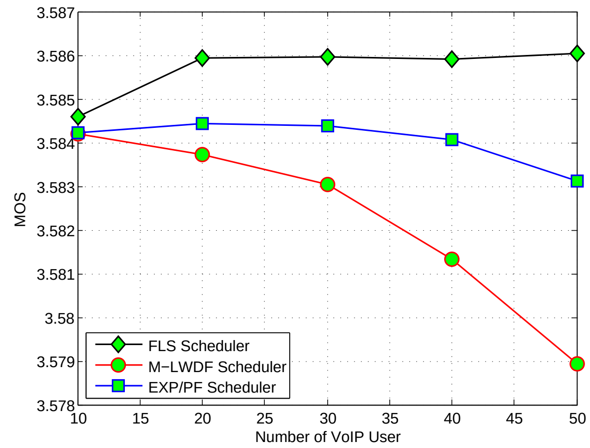

Fig. 3. MOS vs number of VoIP user with AMR-WB mode 0

scheduler continue to keep the first position of the lowest MOS and the EXP/PF scheduler is in the middle of the others. However in this case, the FLS scheduler seems to slightly decreases when the NU increase. This is quite different from the case of AMR-WB mode 0. M-LWDF scheduler still trends to heavily decrease MOS score when the NU increases.

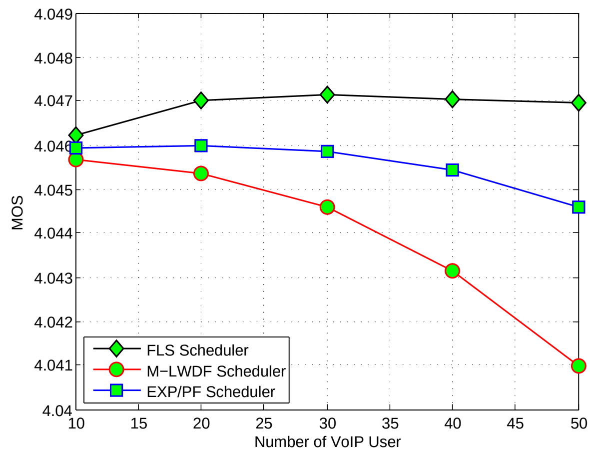

Fig. 4. MOS vs number of VoIP user with AMR-WB mode 1

For AMR-WB anchor bitrate (i.e. AMR-WB mode 1) such as represented on Figure 5, it is also similar to the AMR-WB mode 0 and 1, in this case, the FLS scheduler still has the highest MOS score while the M-LWDF continues to have the lowest MOS score. It is really the difference among the schedulers is not too much, the most clear difference is the trends of them. For example, the FLS scheduler trends to quite slightly, the EXP/PF scheduler trends to averagely reduce while the M-LWDF scheduler trends to heavily decrease when the NU increases.
In order to evaluate the effects of AMR-WB modes on voice quality, we calculate the average MOS score for all modes such as represented on Figure 6. It is clear that, in general , the average voice quality decreases when the NU increases. The principles of the schedulers for AMR-WB mode 0, 1, and 2 are still suitable for this case. It can be concluded that FLS scheduler always has the best voice quality, M-LWDF scheduler has the worst one while EXP/PF is in the middle of two remaining schedulers.

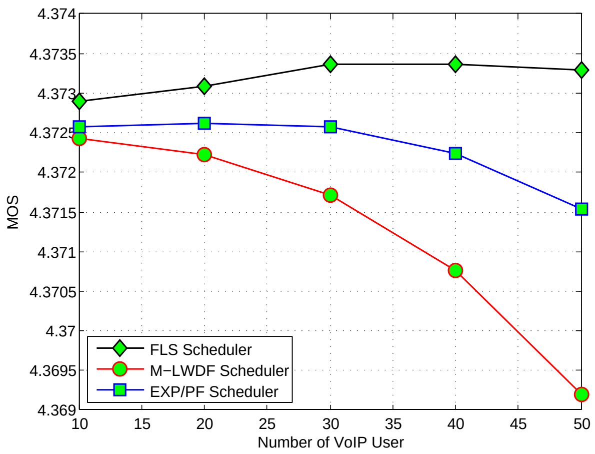

Fig. 5. MOS vs number of VoIP user with AMR-WB mode 2

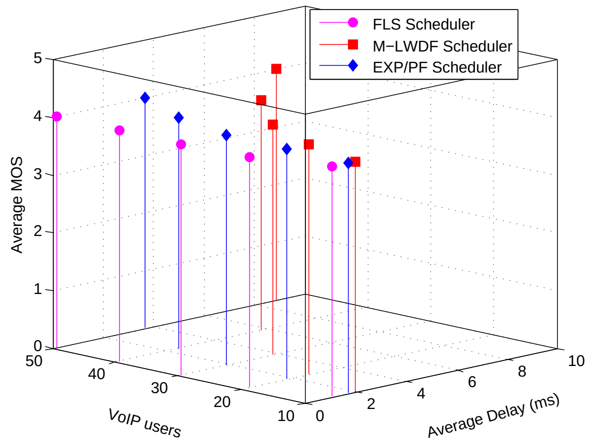

Fig. 7. Effects of Delay and the number of VoIP user on voice quality

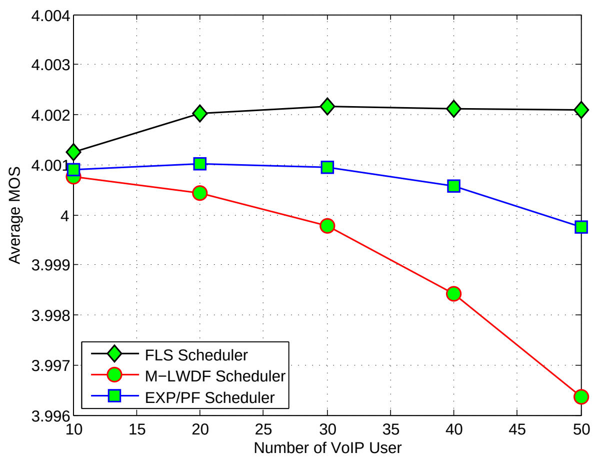

Fig. 6. Average MOS vs the number of VoIP user

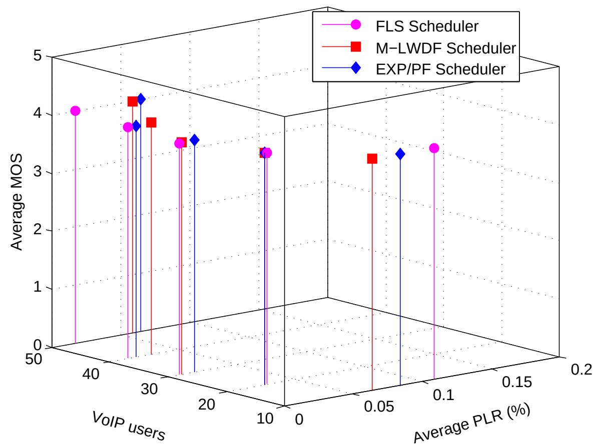

Fig. 8. Effects of PLR and the number of VoIP user on voice quality

4.2.2 Effects of Delay on voice quality. In order to assess simultaneously the effects of delay and the NU on voice quality, we present the relationship between the delay, the NU with the MOS score such as represented on Figure 7. Both M-LWDF and the EXP/PF schedulers have the average delay increasing when the NU increase, thus, MOS score decreases. This means the higher NU and the higher delay, the lower user perception. For the FLS scheduler is quite special, the delay decreases when the NU increases, thus, the MOS score slightly increases. This is unprecedented. Maybe it reflects the stable in a real system because the LTE-Sim is quite similar to a real system.

4.2.3 Effects of PLR on voice quality. For the effects of PLR and the NU on voice quality such as shown on Figure 8. All the schedulers have the stable PLR when the NU increases. However, for the average MOS, the FLS scheduler has the best performance, the M-LWDF scheduler has the worst performance while the EXP/PF is in the middle of two remaining others.
The details of the simulation results of effects of average delay and average PLR according to the NU are shown in Table 6.

Table 6. The detailed results of Figures of 7 - 8

NU	FLS			M-LWDF			EXP/PF		
	Avg. Delay (ms)	Avg. PLR (%)	AVG. MOS	Avg. Delay (ms)	Avg. PLR (%)	AVG. MOS	Avg. Delay (ms)	Avg. PLR (%)	AVG. MOS
10	9.51	0.13	**4.0012**	1.95	0.066	**4.0008**	1.75	0.128	**4.0009**
20	8.83	0.012	**4.002**	2.5	0.058	**4.0004**	1.76	0.041	**4.001**
30	8.34	0.011	**4.0022**	3.43	0.008	**3.9998**	1.85	0.028	**4.0009**
40	8.04	0.015	**4.0021**	5.29	0.041	**3.9984**	2.3	0.002	**4.0006**
50	8.24	0.004	**4.0021**	8.02	0.048	**3.9964**	3.24	0.049	**3.9997**

5. CONCLUSION

In this paper, we propose a new non-intrusive model for predicting voice quality over LTE network. The proposed method is the combination of the LTE-Sim framework and the Wideband E-model. Besides, we also propose to integrate the AMR-WB codec into LTE-Sim because this vocoder is mandatory for VoLTE. The Wideband E-model can predict not exactly VoLTE quality such as intrusive methods (e.g. PESQ-WB), but it does not require original signals to refer, thus, it is very suitable for real-time services such as VoLTE. The advantages of the proposed model is that it is simple, calculate quickly, not much time consuming, and can be applied to many difference simulation scenarios which can be configured in the LTE-Sim. It can be used well for purposes of transmission planning as well as voice quality assessment in laboratory for academic community and researchers. The simulations results show that user perception is quite good when the MOS scores of all the cases are more than 3.5 which is a threshold for almost VoIP users accepted. The simulation results show that the FLS scheduler has the best user perception, the M-LWDF scheduler has the worst one while the EXP/PF scheduler is the middle of two remaining ones. However, it can be seen that, the differences of user satisfaction among them are not significant. The user perceptions of three schedulers increase when the mode of AMR-WB codec increases. This is too logical. It can be concluded that all of three schedulers are very suitable for VoLTE as well as other VoIP applications. In the near future, we will evaluate the proposed model for more the NU, other voice codecs, and complement other essential parameters into the Wideband E-model.

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