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Performance Evaluation of QoS for VoIP and Video Streaming over LTE Networks

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Abstract: With the newest multimedia services supported to the user under existence of the Fourth Generation (4G) of wireless communications networks, Long Term Evolution (LTE) networks have succeeded to achieve a higher user satisfaction factor. Quality of Services (QoS$_s$) is a major focus by the User Equipment (UE) that impacts significantly on the Quality of Experience (QoE) and can be used to measure the user satisfaction factor of the multimedia services. QoE is a new principle that analyzes the QoS through taking user knowledge into account. Voice over IP (VoIP) and Video Streaming are common to be highly used by users all the time. Therefore, for VoIP, Mean Opinion Score (MOS) and impairment factor due to total losses and end-to-end delay are estimated while, for Video Streaming, Packet Loss Rate (PLR), average throughput and Fairness Index are estimated. Channel conditions are considered into this paper to help in the assessment for multimedia quality. Transmission rate (R-factor), end-to-end delay and total losses percentage are factors which impact on VoIP quality while number of users is impressive on Video Streaming quality. MATLAB simulation program can be utilized to check the validation of the system model. Simulation results are introduced and discussed well.

Keywords: LTE, UE, QoS, QoE, delay, losses, Voice over IP (VoIP), Video Streaming and Proportional Fair (PF)

I. Introduction

Day by day, multimedia services through wireless technologies have become more worldwide usable over the past decade due to the numerous developments in the area of wireless technologies. Since the past decades, the evolution of wireless communications has been based on the growing demand of users for multimedia services like (e.g., Facebook, Skype, G-Talk, Online Gaming, Web applications, Video Streaming, VoIP and etc) [1,2]. Therefore, user's satisfaction factor has become a great scope that should be achieved over the wireless communications networks [1,3].

Regarding to modern User Equipment (UE) devices, multimedia services and streaming are provided to support the huge demand of users especially in Long Term Evolution (LTE) wireless communication technology. The Third Generation Partnership Project (3GPP) has developed the Universal Mobile Telecommunications System (UMTS) of the Third Generation (3G) wireless communications into the Long Term Evolution of the Fourth Generation (4G) wireless communications to satisfy more benefits [14], one of them is the higher data rate as 100 Mbps (for LTE) up to 1 Gbps (for LTE-Advanced) [4,5,14]. In order to achieve the user's satisfaction factor, Quality of Services (QoS$_s$) is an important issue to be analyzed [1,6,8,11,12], what has its vital effect on the Quality of Experience (QoE) by the UE [1,11]. QoS includes such parameters which they describe the performance of the services supported to the user. These parameters are: Mean Opinion Score (MOS), End-to-End delay, jitter, Packet Loss Rate (PLR) and throughput. All these previous parameters change the QoS and consequently have their impact on the QoE. QoE comes from the user's experience of noisy channel, empty gaps and background [1,11].

VoIP and Video Streaming are very common of the multimedia services which users can acquire them [1,6-12]. In VoIP traffic model, MOS is an important parameter to evaluate the QoS for voice over Internet Protocol (IP) and it is known that MOS varies from one score to five scores in ascending quality [1,7,8]. There are two basic methods to measure the QoS for VoIP: Subjective and Objective methods [1,7]. MOS is an epitome for Subjective method while the Objective method utilizes various models of user prediction to obtain a better performance for VoIP in an automatic process without human interference [1,3,7,8]. Losses and delay are considered in the performance analysis of QoS for VoIP. In Video Streaming, LTE system has introduced a well approach for this provided service [10]. Number of video streaming users is a main factor in the performance analysis of QoS in this paper since it impacts on the throughput and PLR. Fairness of video streaming resources is assigned among number of users [10]. Channel conditions such as path loss, path loss exponent according to environment, Signal to Interference plus Noise Ratio (SINR), achievable user's data rate and modulation techniques are considered in the numerical analysis for QoS performance evaluation.

The rest of the paper is organized as follows: Section 2 explains a quick review on the multimedia services over UMTS and Section 3 introduces VoIP and Video Streaming multimedia QoS over LTE. Section 4 provides the system model and mathematical expressions of the study and Section 5 presents the simulation results and discussion. Section 6 concludes this paper and predicts with the future work.

II. Multimedia Services over UMTS

UMTS networks are evolved from the Third Generation of mobile wireless communications (3G) [14]. They have supported a great job in mobile broadband, packet based text transmission, encoded voice, video applications and multimedia services [14,16]. These applications and services are supported by UMTS networks at data rates reaching up to 2 Mbps [14]. End users are the most beneficiary of UMTS services. Regarding to Internet access, when UMTS is completely available, end users can access the Internet even though they are travelling and roaming. One of the most important drawbacks what face UMTS is the complexity of connections used for Internet access but UMTS could overcome this problem and make it easier [14,15]. It could be concluded that UMTS services are interested in alternative billing methods or calling plans [14,15,16]. Larger bandwidth of UMTS supports new other services such as Video Conferencing and IPTV [15].

III. VoIP and Video Streaming QoS over LTE

According to the previous section, Section 2, LTE has enhanced multimedia services quality especially for the common both services, VoIP and Video Streaming. The enhancement is coming from the higher downlink data rate of LTE compared to UMTS. From 100 Mbps (for LTE) up to 1 Gbps (for LTE-A) [1,11,14], it is the downlink data rate that supports higher QoS improving the UE QoE. VoIP QoS over LTE has been measured through QoS parameters that are mentioned in the introduction section. One of these parameters is the MOS. MOS in LTE is measured by the Objective method utilizing the E-Model rather than the Perception Evaluation of Speech Quality (PESQ) [1]. However the PESQ is an Objective method, it could not be used to monitor the QoE for the Real Time calls (RT calls) since it uses a reference signal and associates it with the real attenuated signal to obtain the MOS result [1,8]. Consequently, E-Model is used to estimate the MOS score of voice quality depending on the Transmission Rate factor (R-factor) [1,7,8]. Moreover, Impairment Factor due to total losses and End-to-End delay are estimated. Video Streaming over LTE can be characterized by the source data rate and UE speed in the target area [10]. Throughput, PLR and Fairness are important parameters describe video streaming QoS. Data traffic, either VoIP or Video Streaming, is performed through a packet scheduling algorithm between the serving buffer at the eNodeB (4G Base Station) and the UE_s as shown in Fig. 1.

Transmitted power of eNodeB, path loss model, path loss exponent for specific environment, cell radius, spectral power density, channel bandwidth, operating frequency and noise figure by UE are taken into account for SINR calculation. According to the channel bandwidth value, number of Resource Blocks (RB) can be determined as each RB contains twelve data subcarriers [4,10] (data may be VoIP or Video Streaming). Each two consecutive RB_s are transmitted over the Transmission Time Interval (TTI) from the eNodeB to the UE [4,10] as illustrated in Fig. 1. As well as the latter, the achievable user's data rate is well estimated.

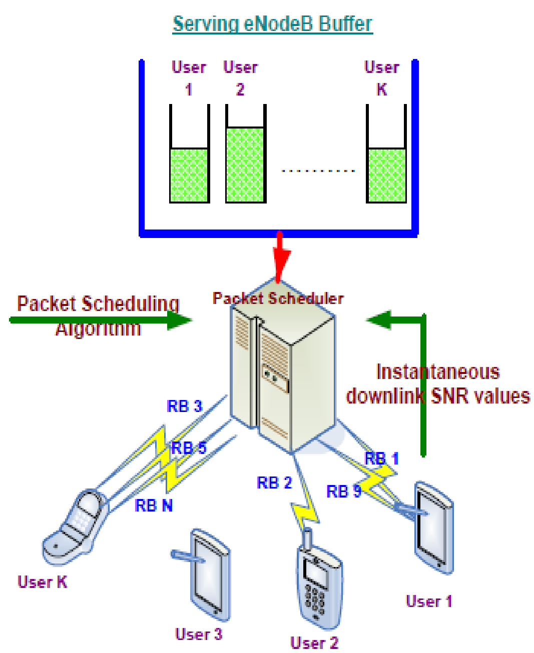

Fig. 1: 3GPP Downlink Packet Scheduling General Model [10]

IV. Proposed System Model

In the proposed system model, two real-time traffic models are presented, VOIP and Video Streaming, QoS performance evaluation metrics. Moreover, channel and environment condition parameters are analyzed to determine the impact on quality. The system model may be divided into four sub-sections as the following:

VoIP Traffic Model

By applying one of the Objective method techniques [1] to measure voice over IP quality, E-Model using R-factor, MOS can be derived as the following:

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$$MOS = \begin{cases} 1 & R < 1 \\ 1 + 0.035R + R(R-60)(100-R)7.10^{-6} & 0 < R < 100 \\ 4.5 & R > 100 \end{cases} \quad (1)$$

In equation (1), it can be obvious that MOS scores didn't reach to 5 (the best speech quality rating). This is due to the objective methodology of the E-model technique by using R-factor which doesn't depend on human interaction for speech quality ratings. Delay and losses are considered in the automated measurement by the UE.

R represents the transmission factor (R-factor) derived in [1] in the simple form as follows:

$$R = 94.2 - I_d - I_{ef} \quad (2)$$

Where: I_d : impairment factor due to one way delay (end-to-end delay) in msec; I_{ef}: effective equipment impairment factor due to total losses (including network and buffer losses), and 94.2: acts the calculation of ($R_o - I_s$+A), where R_o: basic SNR; I_s: mixture of overall impairment factors due to VOIP quality, and A: advantage factor

Impairment Factor due to End-to-End delay (including the delay occurred from the source to the destination) can be described as the following expression:

$$I_d = 0.024d + 0.11\,(d - 177.3)H(d - 177.3) \quad (3)$$

Where: d: End-to-end delay in msec and $H(x)$: Step function

Step function $H(x)$ can be expressed as follows:

$$H(x) = \begin{cases} 1 & if\ x \geq 0 \\ 0 & if\ x < 0 \end{cases} \quad (4)$$

And, Effective Equipment Impairment factor I_{ef} due to total losses (Hint: total losses = network losses + buffer losses) can be expressed as follows:

$$I_{ef} = \gamma_1 + \gamma_2 * \ln(1 + \gamma_3\, e) \quad (5)$$

Where: I_{ef} : effective equipment impairment factor caused by total losses; γ_1 : voice quality impairment factor due to the encoder; γ_2, γ_3 : impact of loss on voice quality for a specific codec, and e: total losses percentage % that includes losses of network and losses of the buffer which caused by waiting delay

Video Streaming Traffic Model

A Video Streaming session is acted as the entire Video Streaming real time traffic. As shown in Fig. 2, each video streaming session has a simulation time. This simulation time changes from 0 to KT where K refers to number of frames which the session contains and T belongs to the arrival time of each frame. Each frame of video session arrives at a regular interval T based on the arrival rate of frames per second. Each frame consists of a fixed number of packets, each packet is transmitted. These packets are distributed according to Truncated Pareto [4,10,11]. There is an interval time between each two arriving packets, called inter-arrival time between each two packets. This delay occurred between each two packets is due to encoding delay and Dc at the video encoder [11].

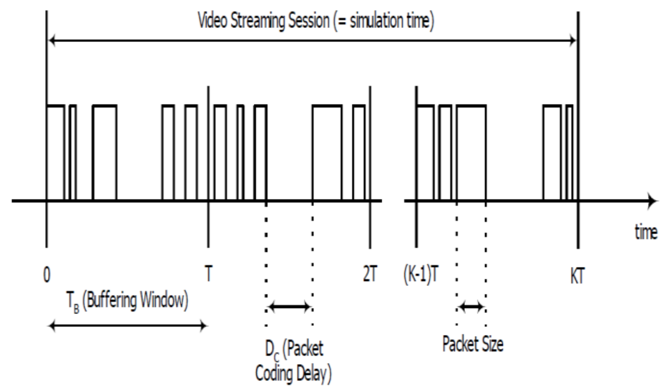

Fig. 2: Video Streaming Traffic Structure [4]

TB is the length (in seconds) of the buffering window in the UE so that it can guarantee a continuous display of video streaming data [4,10]. This parameter is useful for identifying periods when the real-time constraint of this service is not met [4]. Transmitted packets are scheduled on every TTI in the form of RB_s through a packet scheduling algorithm as observed from Fig. 1. Qos of Video Streaming multimedia service can be evaluated according to these following parameters: average system throughput, PLR and fairness index. The average system throughput is estimated as the following function:

$$System\ throughput = \frac{1}{KT} \sum_{i=1}^{N} \sum_{t=1}^{KT} P_{ti}\,(t) \quad (6)$$

Where: $P_{ti}(t)$: size of transmitted packets for user i at time t and N: number of Video Streaming users

$$PLR = \frac{\sum_{i=1}^{N} \sum_{t=1}^{KT} P_{losti}(t)}{\sum_{i=1}^{N} \sum_{t=1}^{KT} P_{totali}(t)} \quad (7)$$

Where: $P_{losti}(t)$: size of lost packets for user i at time t and $P_{totali}(t)$: size of total packets arriving to the serving eNodeB buffer for user i at time t

$$Fairness = 1 - \frac{(P_{tmax} - P_{tmin})}{\sum_{i=1}^{N} \sum_{t=1}^{KT} P_{totali}(t)} \quad (8)$$

Where: P_{tmax} : size of total transmitted packets of the highest served users and P_{tmin}: size of total transmitted packets of the lowest served users

Signal to Interference plus Noise Ratio

Assuming all users are positioned at the edges of the cell [10], the gain is constant for all users since the distance between BS and users is the cell radius. Therefore, the downlink $SINR$ is derived as the following expression:

$$SINR(d) = \frac{P_{out}}{P_L(d) F N_o B} \quad (9)$$

Where: P_{out}: output power from the eNodeB in Watt; $P_L(d)$: path loss quantity in dB; F: noise figure by UE in dB; N_o: spectral noise density in dBm/Hz, and B: channel bandwidth in MHz

Path loss is modeled by large scale model that is known with “One-Slope” path loss model. One-Slope path loss model can be calculated as the following function:

$$P_L(d) = P_L(d_o) + 10n\log(\frac{d}{d_o}) \quad (10)$$

Where: $P_L(d_o)$: free space path loss quantity at reference distance d_o in dB; n: path loss exponent in dB, and d: separation distance between BS and UE in meters

Achievable user’s data rate $\boldsymbol{r_i(t)}$

According to scheduling algorithm [1,4,10,13,17], each two consecutive RB_s are sent from the packet scheduler to user i at every TTI. Each user i has an achievable data rate that can be expressed as follows:

$$r_i(t) = \frac{N_{bits}}{symbol} * \frac{N_{symbols}}{slot\ time} * \frac{N_{slots}}{TTI} * \frac{N_{sc}}{RB} \quad (11)$$

Where: N_{bits}: number of bits per symbol (variable according to modulation technique); $N_{symbols}$: number of symbols per slot time; N_{slots}: number of slots per TTI, and N_{sc}: number of subcarriers per RB

V. Simulation Results and Discussion

Experiment Setup

In this section, the proposed system model can be simulated by using MATLAB R2010b simulation program. QoS for VoIP traffic model can be estimated according to the following factors: MOS via E-Model by using R-factor, impairment factor caused by one way delay (end-to-end delay) I_d and effective equipment impairment factor due to total losses I_{ef}. Table 1 indicated VoIP traffic model simulation parameters. QoS for Video Streaming traffic model may be evaluated to assign its performance through the following factors: average system throughput, PLR and fairness index. Table 2 includes Video Streaming traffic model simulation parameters. Regarding scheduling algorithm, achievable user i data rate $r_i(t)$ can be estimated for LTE networks according to Proportional Fair (PF) [1,11,13,17] scheduling algorithm used in the simulation. Moreover, downlink $SINR$ is calculated for the eNodeB transmission to served users. Table 3 contains channel simulation parameters useful for downlink $SINR$ and $r_i(t)$ calculations.

TABLE 1 VoIP Traffic Model Simulation Parameters

Parameter	Value
R-factor	10 - 90
End-to-end delay, d	0 – 0.4 sec
γ_1	11 for (G.729-A Codec) and 0 for (G.711 Codec)
γ_2	40 for (G.729-A Codec) and 30 for (G.711 Codec)
γ_3	10 for (G.729-A Codec) and 15 for (G.711 Codec)

TABLE 2 Video Streaming Traffic Model Simulation Parameters

Parameter	Distribution	Value
Number of frames per second, K	Deterministic	20
Arrival time of each frame, T	Deterministic	50 msec
Number of packets per each frame	Deterministic	8
Inter-arrival time between each two packets	Truncated Pareto	2.5 msec
Packet size	Truncated Pareto	320 bits
Number of Video Streaming users, N	Uniform	80 - 120

TABLE 3 Channel Conditions Simulation Parameters

Parameter	Value
Channel Bandwidth, B	20 MHz

Cell radius	500 meters
Operating frequency, f	2 GHz
Max eNodeB transmit power, P_{out}	20 W
Number of channel subcarriers	1200
Number of subcarriers per RB	12
Number of RB_s	100
Transmission Time Interval, TTI	1 msec
Slot time	0.5 msec
Number of symbols per slot time	7
Path loss exponent, n	2.7 – 3.5 dB
Path loss model, $P_L(d)$	One-Slope
Noise spectral density, N_o	-174 dBm/Hz
Noise figure by UE, F	9 dB

The simulation output can be assigned as the following:

VoIP Traffic Model

MOS, I_d and I_{ef} can be simulated by using equations (1), (3) and (5) respectively.

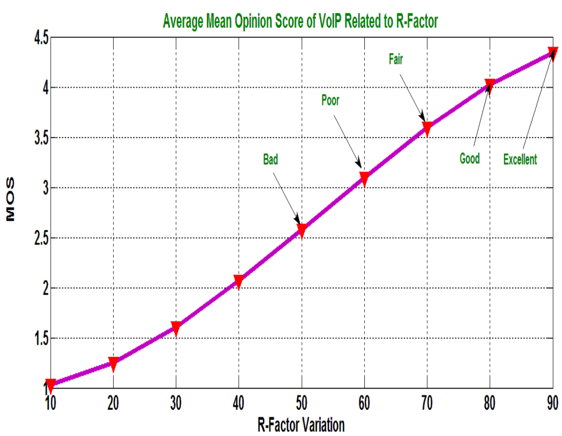

Fig. 3: Mean Opinion Score of VoIP Related to E-Model Using R-factor

Figure 3 depicts the MOS results for VoIP related to R-factor by using the E-Model function expressed in equation (1). As shown in this figure, if R-factor varies from 10 to 90, the MOS increases from above 1 to below 4.5 by using equation (1). Then, it may be observed that transmissions with less than 50 are experienced to provide “Bad” QoS to the user whereas transmissions above 50 up to 60, above 60 up to 70, above 70 up to 80 and above 80 up to 90 are experienced to provide “Poor”, “Fair”, “Good” and “Excellent” respectively. Table 4 indicates the user experience for these transmissions of VoIP traffic model and their impact on the users’ satisfaction.

TABLE 4 QOE for VoIP by E-Model Using R-factor

R-factor	MOS	User Experience	Satisfaction
90	4.34	Excellent	Nearly all satisfied
80	4	Good	Many satisfied
70	3.60	Fair	Some satisfied
60	3.10	Poor	Few satisfied
50	2.58	Bad	Nearly nothing satisfied

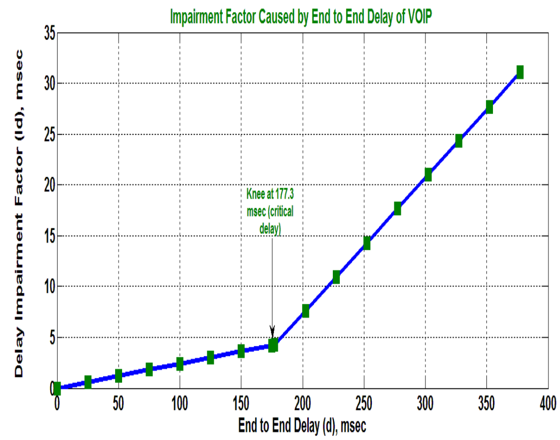

Fig. 4: Impairment Factor of VoIP Related End to End Delay

Figure 4 depicts the impairment factor due to the end-to-end delay I_d of VoIP, end-to-end delay d is the delay happened from the source to the destination. As shown in this figure, end-to-end delay d can be divided into two roughly linear regions. The first liner region is before 177.3 msec since this value is the critical delay represented on the curve in a form of a knee and the conversations over IP occur naturally. The second linear region is after this knee, after 177.3 msec, I_d begins to be increased seriously and the conversations will be strain and may be corrupted.

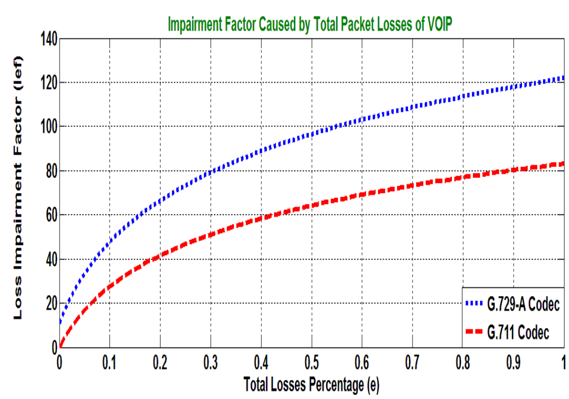

Fig. 5: Effective Equipment Impairment Factor of VoIP Related to Total Losses Percentage

Figure 5 depicts the effective equipment impairment factor due to total losses percentage I_{ef} of VoIP on two different codecs G.729-A and G.711 by using equation (4). In Table 1, there are the parameters for each codec. Total losses percentage e

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includes the losses of network and buffer due to the waiting time of packets at the serving eNodeB buffer. As shown in this figure, if e increases, I_{ef} consequently increases. It can be observed that total losses impact on VoIP is lower in G.711 Codec that that in G.729-A Codec.

Video Streaming Traffic Model

$Average\ system\ throughput$, PLR and $Fairness\ Index$ can be simulated by using equations (6), (7) and (8) respectively.

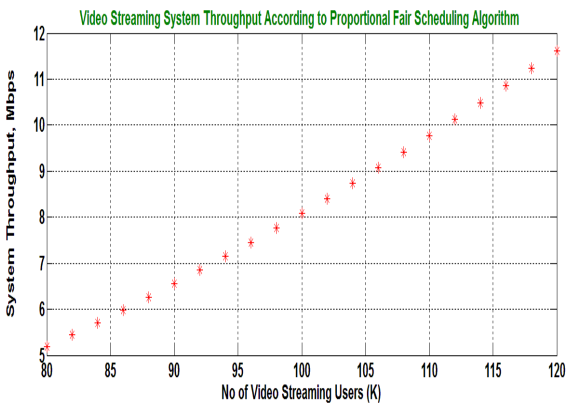

Fig. 6: Video Streaming Traffic Throughput Related to Number of Users

Figure 6 depicts the system average throughput of Video Streaming traffic in a relationship with the number of Video Streaming users N. As known for LTE networks and Proportional Fair (PF) scheduling algorithm, the number of Video Streaming users N ranges from 80 to 120 [10]. In this figure, the simulation considers 128 Kbps source data rate of the Video Streaming traffic. From this shown figure, it can be noticed that if number of Video Streaming users N increases, the system throughput should be increased in order to satisfy user demands in the entire Video Streaming session.

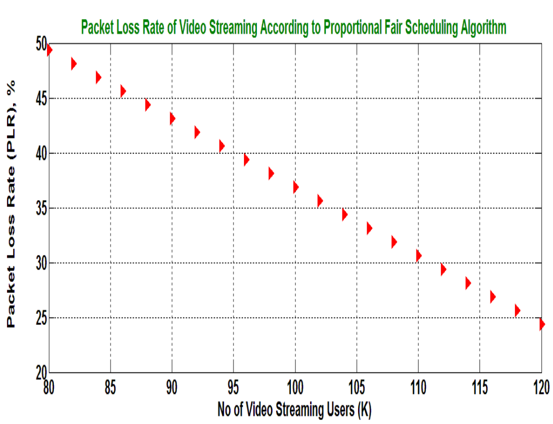

Fig. 7: Video Streaming Traffic PLR Related to Number of Users

Figure 7 depicts the Packet Loss Rate PLR of Video Streaming data traffic in a relationship with the number of Video Streaming users N. As well as the average system throughput of Video Streaming should be increased when the number of users increases, also the Packet Loss Rate PLR should be decreased regarding PF scheduling algorithm. Due to growing number of Video Streaming users N, fairness between UE_s must be achieved. By using equation (8), $Fairness\ Index$ can be calculated and inserted into Table 5 related to number of Video Streaming users N. Table 5 contains accurate simulation results of $Fairness\ Index$ among UE_s.

TABLE 5 Fairness Index Related to Number of Video Streaming Users

N	$Fairness\ Index$
80	0.9939
100	0.9947
110	0.9951
120	0.9959

It can be noticed from Table 5 that when number of Video Streaming users N increases, $Fairness\ Index$ is increased regarding PF scheduling algorithm. It can be concluded that increased throughput leads to decreased Packet Loss Rate and increased Signal to Interference plus Noise Ratio $\boldsymbol{SINR}$.

By using equations (9) and (10) and with the aid of simulation parameters in Table 3, can be estimated.

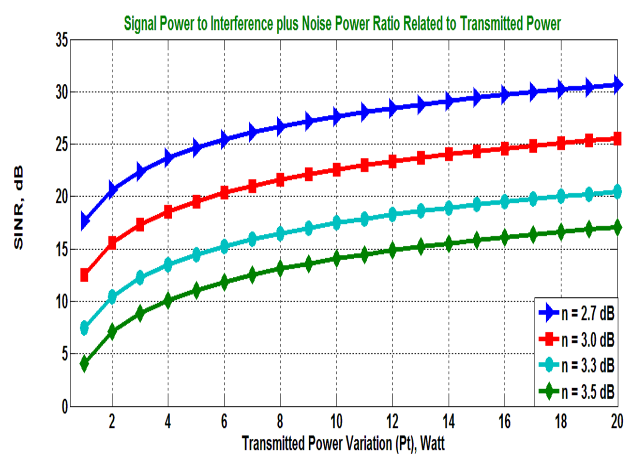

Fig. 8: Signal to Interference plus Noise Ratio Related to Transmit Power

Figure 8 depicts the Signal to Interference plus Noise Ratio related to transmitted power from the eNodeB. As shown in this figure, when P_t increases, SINR logarithmically increases since more transmitted power, more signal power can overcome the total noise power. When path loss exponent increases from range 2.7 dB to 3.5 dB, SINR is decreased due to higher large scale path loss model. SINR result values are ranging from approximate 5 dB to approximate 30 dB.

Achievable user's data rate

By using equation (11) and with the aid of simulation parameters in Table 3, achievable user's data rate can be calculated according to modulation technique that differentiates number of bits per symbol.

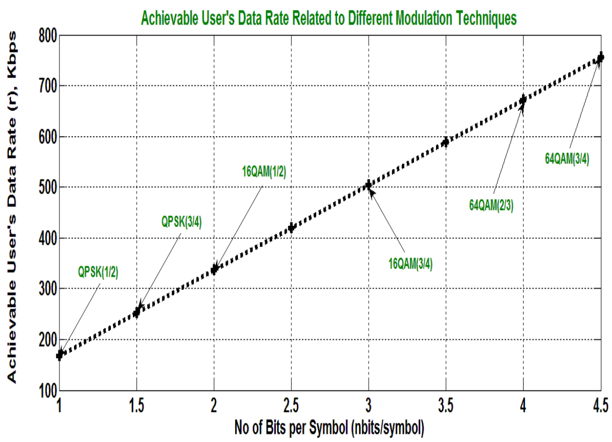

Fig. 9: Achievable User's Data Rate Related to Modulation Technique

Figure 9 depicts the achievable user's data rate related to different modulation techniques. As shown in this figure, when number of bits per symbol increases, the achievable user's data rate increases from 168 Kbps up to 756 Kbps. Modulation techniques with coding rate controls the achievable user's data rate for each user as other terms of equation (11) are constant.

VI. Conclusion

In this paper, Voice over Internet Protocol (VOIP) and Video Streaming real time multimedia services are evaluated analytically to assign their quality over Long Term Evolution (LTE) of 4G communications networks. Quality of Service (QoS) is a vital issue to be estimated so that the User Equipment (UE) can experience the quality of the supported multimedia services. For VoIP traffic model, Mean Opinion Score (MOS) is calculated by using one of the Objective methods to assign speech quality in range from 1 to 5. The E-Model by using R-factor is used to measure the MOS of VoIP. Impairment factors due to end-to-end delay and total losses are estimated. Impairment factor due to end-to-end delay I_d which includes the delay occurred from source to destination is estimated. Effective equipment impairment factor I_{ef} due to losses of network and buffer is obtained through using two different Codecs, G.729-A and G.711, and consequently a comparison is assigned between them regarding total losses percentage e. For Video Streaming traffic model, Proportional Fair (PF) scheduling algorithm is taken into consideration in the downlink data transmission. System average throughput, Packet Loss Rate (PLR) and Fairness Index are calculated to evaluate Video Streaming traffic model QoS. Channel conditions such as: eNodeB radiated power, path loss model, cell radius, path loss exponent, channel bandwidth, operating frequency, spectral noise density, noise figure by UE and modulation techniques are considered for the estimation of Signal to Interference plus Noise Ratio (SINR) and achievable user's data rate at every Transmission Time Interval (TTI). MATLAB R2010b simulation program is used to check the validation of the numerical analysis and proposed system model. Tables 1, 2 and 3 indicate the used simulation parameters. Tables 4 and 5 provide accurate simulation results for MOS of VoIP and Fairness Index of Video Streaming, respectively.

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