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Х. Х. А. Аль-Заяді
Національний університет "Львівська політехніка"

ЗАБЕЗПЕЧЕННЯ ЯКОСТІ СПРИЙНЯТТЯ ПОСЛУГ ТА ПРОПОРЦІЙНОГО РОЗПОДІЛУ РЕСУРСІВ LTE У ПРОЦЕСІ ВІДЕОТРАНСЛЯЦІЇ

Проаналізовано проблему передавання потокового відео по мережі LTE. Запропоновано математичну модель розподілу ресурсів LTE під час трансляції потокового відео. Це забезпечує різний спосіб обслуговування для гетерогенного мобільного трафіку. Цим досягнуті необхідні параметри якості обслуговування для поліпшення якості сприйняття послуги. Метою роботи є поліпшення якості сприйняття користувачем відеоконтенту в разі надання його засобами мережі радіодоступу LTE. У багатошаровій моделі багатоадресної передачі потокового відео оригінальне відео стискається в кілька шарів. Базовий шар містить найважливіші особливості відео, тоді як інші рівні містять інформацію, яка потім може покращити якість відео. Шари можуть бути відображені в різних IP-групах. У разі додавання відповідних груп, приймач може отримати загальну кількість даних з базового шару з певним рівнем покращення. Отже, нижній шар повинен бути забезпечений нижчим рівнем МС (схеми модуляції і кодування), так, що приймачі, призначені у верхні шари, можуть отримати обслуговування від нижнього шару. Залежно від відстані та розподілу трафіку схеми модуляції та кодування різні, а площа комірки може бути розділена на кілька концентричних кіл. Якщо різні рівні МС використовуються в різних шарах відео, то відповідні дані цих шарів повинні бути передані приймачам всередині відповідних кіл. За підсумками досліджень на основі такої моделі встановлено, що у зашумленому каналі QoE зростає дуже зі збільшенням швидкості передавання. Для такого каналу стратегія справедливого розподілу ресурсів є ефективнішою, тому що за рахунок незначного погіршення QoE досягається істотний виграш у показнику справедливості розподілу.

Ключові слова: LTE, якість сприйняття, якість сервісу, пропорційна справедливість, розподіл ресурсів.

Haider Al-Zayadi
Lviv Polytechnic National University

ENSURING QOE AND FAIRNESS OF LTE RESOURCE ALLOCATION DURING VIDEO STREAMING

The problem of video streaming over LTE network is analyzed. The mathematical model of resource allocation of LTE is offered during video streaming. It provides different service schedule for heterogeneous mobile traffic. It allows assuring the demanded quality parameters for better user experience. The goal of the paper is to increase the LTE user experience delivering video content over the radio access network.

Key words: LTE, QoE, QoS, Proportional Fairness, Resource Allocation.

Introduction

For LTE (Long Term Evolution) the concept of QoS (quality of service) system is used, which is developed even for UMTS networks in specification TS 23.107. QoS should provide necessary number of attributes for multi-tiered gradation of users, be compatible with the efficient use of radio resources policies to support the development of independent core network and radio access network, ensure unambiguous definition of quality attributes. On the basis of the common requirements the technical requirements are formulated in standards: quality of service mechanisms operate on the basis of peer model of “User terminal – Gateway Network”, which provides one-to-one mapping of network services and external applications; quality parameters set should be as short as possible to ensure effective quality control while maintaining efficient use of radio resources and asymmetric cross-channel functioning; implementation of quality management techniques based on successive sessions in the process of multi-streaming media transmission; complexity of network protocols and slower network performance due to the introduction of quality of service should be minimized, as well as the volume of signaling data; applications are provided with ability to control quality of service parameters in the process of transmission in various network nodes; QoS system must be dynamic to be able to change QoS throughout active session.

1. Problem statement

Thus, the actual problem is the development of a model of the LTE service architecture that takes into consideration coordinated solution of QoS ensuring in user and control plane and optimal management of radio resources in LTE network.

2. Review of the literature

The problem of ensuring QoS in case of video streaming is widely covered in papers [1-4], but in these papers are not mentioned the coordination of service quality system with optimal management of radio resources in wireless broadband networks. This is due to the fact that authors mostly do not consider the features of wireless transmission environment and radio access systems. These characteristics necessitate separation of functional structure of QoS system into two subsystems: user plane subsystem and control plane subsystem that must operate independently because between them there is joint radio access network and LTE core network [5-9].

Let’s analyze architecture of QoS system and service transmission within such a system with example of connection establishment between the end equipment connected to the user terminal of mobile network and terminal equipment located in the external packet network (Fig. 1). Here are introduced notions of end-to-end service as a sequence of actions between two end-users and, therefore, partial services – defined by their belonging to a certain network components: transmission service between end-user equipment (TE – terminal equipment) and assigned to the user mobile network terminal (MT – Mobile Terminal); transmission service in the LTE network channel (LTE Bearer Service); transmission service in the outer channel – in the external packet networks (External Bearer Service). Thus, there is a multilevel interaction when transmitting services in different network nodes and at different levels.

Service transmission in LTE, according to the network architecture, is considered separately as in the radio access network (Radio Access Bearer Service), where confidential transmission of user data with pre-selected or set as default in advance level of QoS is ensured, so and in basic packet network (Core Network Bearer Service) that can also support different quality of service. Service in the radio access network is implemented as two components: transmission in radio channel (Radio Bearer Service) and radio access service (Access Bearer Service). Implementation of service in the radio channel includes all aspects related to data transmission on radio interface, including segmentation and re-assembly of user packets. In addition, the physical layer (Physical Radio Bearer Service) manages substreams of user data. Mechanism of radio access ensures data transmission between the radio access network and core network on the physical layer (Physical Bearer Service).

Finally, service transmission in the backbone link (Backbone Network Bearer Service) is considered in the functional set of physical and data layer of OSI model (Open Systems Interconnection) and assigned quality of service parameters.

Thus, based on the presented architecture the decomposition of tasks of the quality of service is carried out. It should be noted that this decomposition is horizontal-vertical by levels and segments of network LTE.

In a user plane such functions are designed to support user traffic and signaling to certain restrictions set by the parameters of QoS.

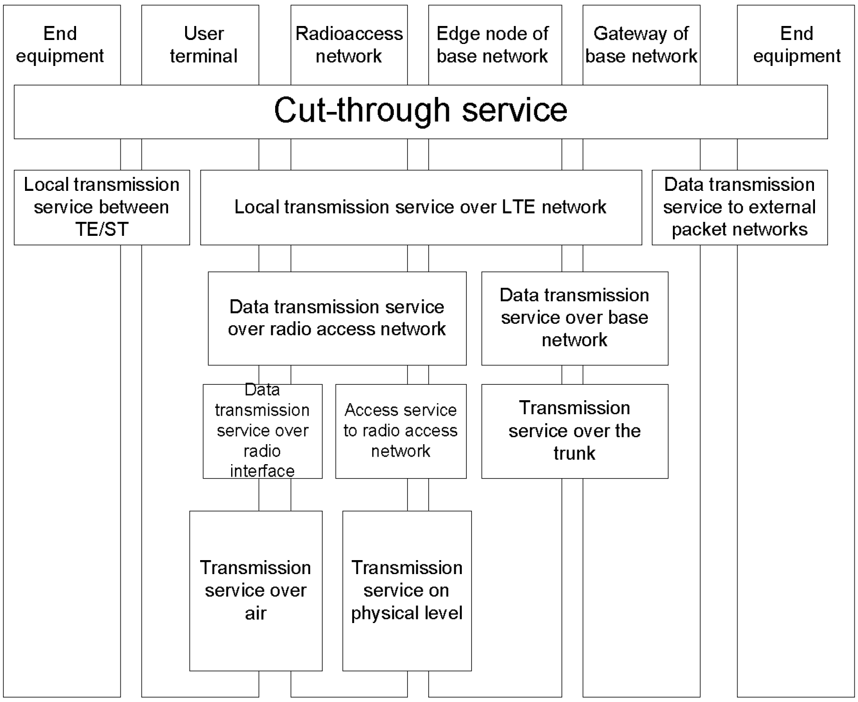

Fig. 1. The functioning of QoS system in the process of connection establishment between the end equipment of LTE network and terminal equipment in the external packet network

Mapping Function (MF) ensures that appropriate parameters of QoS are assigned to each packet for transmission.

Classification function (CF) is intended to define these QoS parameters for packets that are intended for a particular subscriber terminal in case when for this ST few channels for service transmission are established.

Resource Manager Function (RMF) allocates available resources among services according to QoS parameters.

Traffic Conditioner Function (TCF) provides coordination between traffic flow of user data and established QoS level. Those packets that do not meet established QoS parameters, will be rejected or marked as inappropriate for subsequent rejection after accumulation.

Fig. 2 shows interaction of quality of service control function in the user plane.

Classification function that is implemented in UT (user terminal) and SG appoints data packets received from an external (or local) channel into the service of LTE network with appropriate parameters of QoS. Traffic coordination function, when necessary, ensures coordination of the user flow in an upward (in UT) and downward (in SG) directions with established QoS parameters. Next, display function ensures each data packet with QoS-specific indicator and sends it along route over the network that requires allocation of adequate resources. Resource management function is responsible for this and is implemented in each network node.

In the control plane, as always, functions necessary to implement mechanisms of management and control are concentrated.

Service Management Function (SMF) is a coordinating function during installation, modification and management of services and also it controls quality of service for managing functions in the user plane.

Translation Function (TF) converts the internal service primitives of LTE network in modules of different protocols of interacting external networks, including the transformation of attributes of LTE network services in parameters of QoS for external networks protocols.

Admission / Capability Control Function (A / CCF) provides information on all possible resources of network nodes, defining on each request (or modifying) of service, if the network nodes are able to provide the necessary resources. This function also controls ability to provide the service, i.e., whether requested service is implemented in the network.

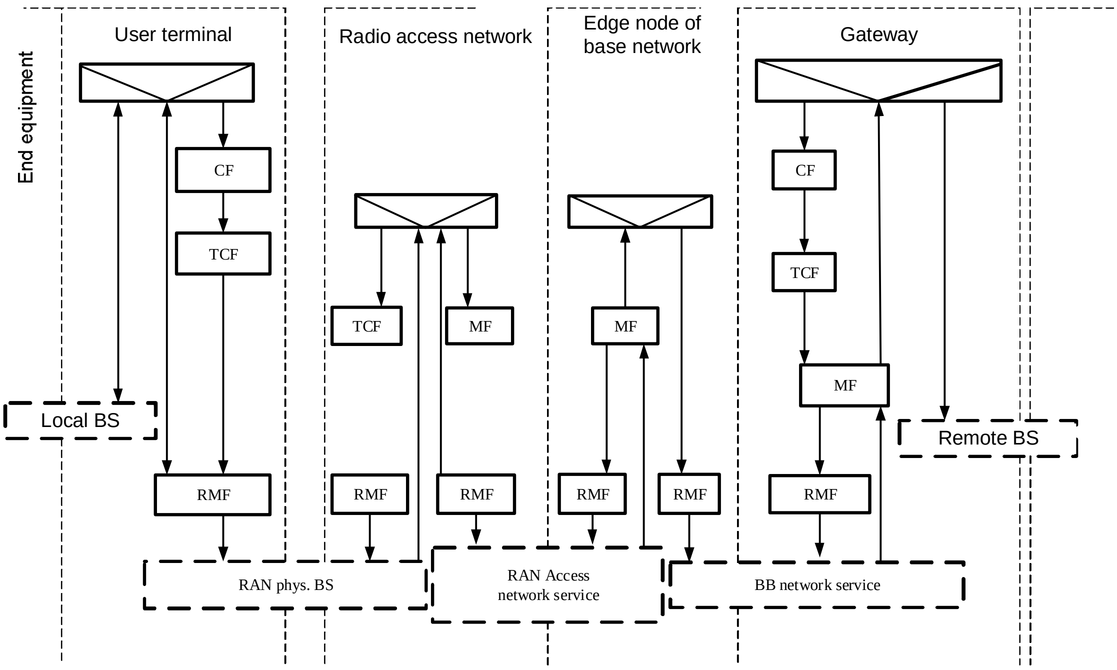

Fig. 2. QoS management function in the user plane

Subscription Control Function (SCF) provides control of availability of different services with the required parameters of QoS for subscribers.

Quality of service management functions interaction in control plane is shown in Fig. 3.

Translation Function, which is operating in the ST and SG, will convert signalling information related to external services in the internal service primitives, including attributes of the service.

Service Management Function, localized in UE, SG and core network (that corresponds to subfunction), using translation function, sets or modifies the service, using the related capability control function to determine resources availability that are needed for this service, and subscription control function in order to determine the user rights to the service.

Note that some QoS parameters are mutually contradictory, such as delay and error rate in the approved package, i.e., reliability. Thus, when transmitting voice traffic cut-through delay should not exceed 150 ms and the allowable loss of information packets should be less than 3%. If we consider stream traffic, in this case the loss of information packets are allowed no more than 1% and for interactive traffic information packet loss generally unacceptable - its services (such as background traffic services) are transmitted in acknowledgement mode and the need for retransmission of accepted with errors packages can't measure the latency.

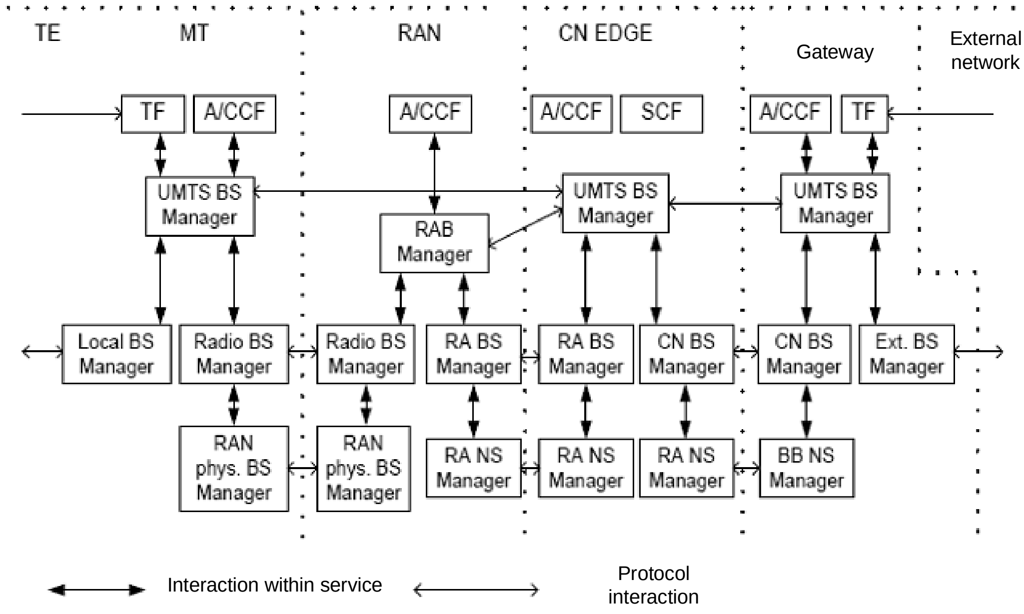

Fig. 3. QoS management function in the control plane

As an illustration we point out the default data transmission parameters when the service is provided in streaming class using AMR (Adaptive Multi Rate) speech codec and MPEG-4 video codec, what is typical for mobile networks of 3^{rd} and 4^{th} generations.

AMR codec:

- data transfer rate 4,75…12,2 kbit/s;
- duration of encoded speech frames 20 ms;
- delay does not exceed 100 ms;
- relative level of bit errors $10^{-2}…10^{-4}$.
- relative level of human error 10-3;

MPEG-4 Video Codec:

- data transfer rate 24,0…128,0 kbit/s;
- total 150…400 ms delay (between edge nodes), including actual video codec delay of about 200 ms;
- relative level of bit errors 10^{-3} (limited use), 10^{-4} (some visible artifacts), 10^{-5} (a small deterioration in perception), 10^{-6} (no visible deterioration).

3. Development of multi-layer model of group video streaming in LTE networks

In a multilayer model of the multiaddress video streaming, original video is compressed into several layers. The base layer contains the most important features of the video, while one or more of the higher levels contain information that can later improve video quality. Layers can be displayed in various IP-groups. By adding relevant groups, the receiver can get the total data from the base layer with a certain level of improvement. Thus, the bottom layer should be assigned lower than MC level (modulation and coding), so that receivers assigned to the upper layers can get service from the lower layer.

Depending on the distribution distance, basic modulation and coding types are different, and the cell area can be divided into several concentric circles. If different MC levels are used in the different video layers, video data of these layers must be transmitted to receivers within the respective circles. Table 1 presents the modulation and coding types corresponding to the relevant sectors of the cell. We consider only the distant zone in the current work.

Let us suppose that S is symbol rate (symbols/s), which is allocated in advance for group video session, there are L various MC levels. Let the possible number of video layers in this system be not more than L and separate MC be applied to each video layer .., $l = 1, 2, ..., L$.

Table 1

LTE modulation and coding

MC Level	Modulation	Size code	bits/ symbol	SNR (dB)	Distance (km)
1	QPSK	1/2	1	3.5	1.58
2	QPSK	3/4	1.5	5.5	1.44
3	16QAM	1/2	2	7.5	1.31

Let us assume that in the outer cell circle where MC level coverage is 1, there are N receivers that are uniformly distributed throughout the cell. Let us assume there are n_l receivers that can be placed inside MC level coverage l, but not within the MC range $l+1$. Let us denote:

$$w_l = \frac{n_l}{N}, \quad where \sum_{l=1}^{L} w_l = 1, \tag{1}$$

S_l – symbol rate allocated for the layer l.

$$S_l = a_l S, \quad where \sum_{l=1}^{L} S_l \le S . \tag{2}$$

Let us define the resource allocation vector A.

$$A = (a_1, a_2, ... a_L), \quad where \sum_{l=1}^{L} a_l \le 1. \tag{3}$$

If the MC level l can modulate r_l bits with one symbol, video-layer l is encoded and transmitted in the band:

$$B_l = S_l \cdot \mathrm{r}_l = a_l \cdot \mathrm{r}_l \cdot S . \tag{4}$$

Resource allocation vector A describes that each MC level is allocated limited resources, so it determines the number of video layers and their transmission rate, i.e. the speed of video encoding.

Receiver within the MC level l coverage area, but not within the level $l+1$, can receive data from video layer 1 to video layer l. We consider the ideal case with no loss, no environment noise where the total data rate R_l of the receiver is:

$$\begin{aligned} & R_l = \sum_{i=1}^{l} B_i = S \sum_{i=1}^{l} a_i \cdot r_i = S \cdot x_l \\ & where x_l = \left\{ \begin{matrix} 0, \ if \ l = 0 \\ x_{l-1} + a_i \cdot r_i, \ for \ 1 \le l \le L \end{matrix} \right\} . \end{aligned} \tag{5}$$

3.1. Balancing between throughput and resource allocation fairness

Resources are allocated to maximize overall throughput. To do this, we have to determine the allocation vector $A_{\max\_thr}$ to maximize the amount of received data rates for all receivers.

$$\begin{aligned} A_{\max\_thr} &= \arg\max_A \left(\sum_{i=1}^{N} R_i\right) = \arg\max_A \left(\sum_{i=1}^{L} n_l \cdot R_i\right) \\ &= \arg\max_A \left(\sum_{i=1}^{L} w_l \cdot R_i\right), \ \text{subject\_to}\left(\sum_{i=1}^{L} S_i \le S\right) \end{aligned} . \tag{6}$$

The proportional fairness (PF) was introduced in game theory, which is the most common method for network planning to establish a compromise between efficiency and consistency of resources, the purpose of which is to maximize the use function. If the use function is the identity function of obtained data rate, we can get the following expression:

$$
\begin{aligned}
A_{\max\_PF thr} &= \arg\max_{A}(\prod_{i=1}^{N} R_i) = \arg\max_{A}(\frac{1}{N}\prod_{i=1}^{N} R_i^{n_l}) \\
&= \arg\max_{A}(\sum_{i=1}^{L} n_l \cdot \log(R_i)) \\
&= \arg\max_{A}(\sum_{i=1}^{L} w_l \cdot \log(R_i)),\ \text{subject\_to}(\sum_{i=1}^{L} S_i \le S)
\end{aligned} \quad (7)
$$

3.2. QoE model

S-curve chosen as a model that summarizes relationship between QoE (Quality of Experience) and distortion, display function QoE reflects the average rate of the video stream *R*, where video encoding rate is indicated as R_S

$$
Q(\mathrm{R}) = \frac{1 - e^{-c_1(\frac{R}{R_s})^{C_2}}}{1 - e^{-c_1}}. \quad (8)
$$

Environment variables C_1 and C_2 indicate video distortion. If $C_1 = 6$, $C_2 = 2$, as a rule, it is assumed that the environment is an ideal wireless channel, and if $C_1 = 6$, $C_2 = 6$, we suppose that this is a noisy channel area.

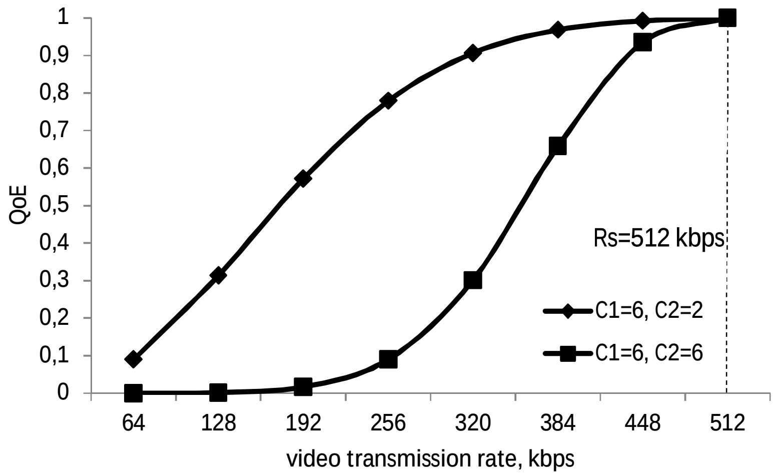

Fig. 4. QoE function

3.3. The optimal and uniform QoE allocation

Allowable quality of users' services experiences is a function of allowable data rate *R* and resource allocation strategy to maximize the total QoE for all receivers:

$$
\begin{aligned}
A_{\max\_QoE} &= \arg\max_{A}(\sum_{i=1}^{N} Q(R_i)) = \arg\max_{A}(\sum_{l=1}^{L} n_i \cdot Q(R_i)) \\
&= \arg\max_{A}(\sum_{i=1}^{L} w_l \cdot Q(R_i)),\ \text{subject\_to}(\sum_{i=1}^{L} S_i \le S)
\end{aligned} \quad (9)
$$

If we consider the uniform allocation between the receivers, min-max allocation fairness demonstrates better quality of experience than PF planning. However, in respect of QoE PF planning provides a practical solution for compromise between allowable quality and QoE of the user.

$$A_{\max\_PFQoE} = \arg\max_A(\prod_{i=1}^{N} Q(R_i)) = \arg\max_A(\frac{1}{N}\prod_{l}^{L} Q(R_i)^{n_l})$$
$$= \arg\max_A(\frac{1}{N}\sum_{i=1}^{L} n_l \cdot \log(\mathrm{Q}(R_i))) \tag{10}$$
$$= \arg\max_A(\sum_{i=1}^{L} w_l \cdot \log(\mathrm{Q}(R_i))),\ \text{subject\_to}(\sum_{i=1}^{L} S_i \le S)$$

3.4. Assessment of QoE allocation fairness

Let us assume that Q denotes the average QoE of N receivers, fairness factor then can be defined as follows:

$$F(Q(\mathrm{R}_1), Q(\mathrm{R}_2), ..., Q(\mathrm{R}_N)) =$$
$$= 1 - \frac{1}{2(\mathrm{N}-1)}\sum_{i=1}^{N}\left|\frac{Q(\mathrm{R}_i)}{Q} - 1\right| \tag{11}$$
$$= 1 - \frac{N}{2(\mathrm{N}-1)}\sum_{l=1}^{L} w_l \cdot \left|\frac{Q(\mathrm{R}_i)}{Q} - 1\right|$$

4. QoE parameters research during the video traffic transmission with limited radio network resources

Performance analysis and simulation were performed based on the following assumptions:

- eNodeB provides 3 types of modulation and coding schemes for distant zone and their application areas described in Table 1;
- Mobile stations are evenly distributed within the cell;
- Video is encoded using SVC (scalable video coding) at encoding rate R_s*=512 kbps*;
- Number of video layers is no more than the number of modulation and coding scheme types;
- Losses at Scalable Video layers are not accounted.

When using the point-to-point connection, the appropriate MC level for video information transmission is determined depending on the channel condition towards the receiver. For this reason, receivers with low signal / noise ratio at the output choose relatively low MC level and limit speed of multiaddress transmission. Due to this, the receivers receive a fair share of bandwidth, but it does not ensure sufficient service QoE in many cases.

On the other hand, video streaming effectiveness is not measured with bandwidth, as in case of usual data transfer. Hence, video streaming effectiveness depends on the services quality of experience (QoE).

Table 2 and 3 summarize the best results of resource allocation with maximized and proportional fair QoE according to different symbol rates and number of video layers.

Table 2

Optimal resource allocation results with maximized QoE for different symbol rates and amounts of video layers

Video symbol rate (ksym/s)	Video layer	Coding rate	MC type	Symbol rate	QoE	Users, %	Average QoE	Fairness level
120	1	240	3	120	0,91	68	0,62	0,67
200	3	32	3	16	0,9	68	0,83	0,94
	2	63	2	42	0,82	14		
	1	142	1	142	0,56	18		
280	3	22	3	11	0,97	68	0,97	0,99
	2	30	2	20	0,95	14		
	1	249	1	249	0,92	18		

For example, if the available symbol rate is 200 ksym/s, for its streaming we form group multi-layer session and get the maximum average QoE equal to 0.83, provided that:

- The number of video layers is 3;
- Base layer coding rate is 142 kbps and improvement layers coding rates are 63 kbps and 32 kbps, respectively;
- Symbol rate for the base layer is set at 142 ksym/s and first MC type is used (e.g. QPSK modulation and coding rate 1/2);
- For the first improvement layer, symbol rate is set at 42 ksym/s and 2nd MC type is used (i.e., QPSK and coding rate 3/4);
- For the second improvement layer, symbol rate is set at 16 ksym/s and 3d MC type is used (i.e., 16QAM and coding rate 1/2);
- 18% receivers receive only the base layer with 0.56 QoE, 14% receivers receive the base layer and the first enhancement layer with 0.82 QoE while 68% receivers receive all layers with 0.9 QoE.

As compared with the case of QoE maximization, PF QoE allocation strategy encourages video stream sharing by a wider range of users through the division of the video stream to a greater number of layers, and thus improving the quality of fairness, as shown in Table 3.

In case of bandwidth limitation to 120 ksym/s, QoE maximization strategy forms a single layer video stream and covers 68 of users only. And we are using 3d MC level. On the other hand, PF algorithm divides the resources into 3 layers and covers 100% of users by adapting MC of the lower levels to layers 1, 2 and 3 to get more coverage. Therefore, each of 18% users receives video from layer 1 at 80 kbps data rate, each of 14% users receives data from both layers 1 and 2 at an aggregate data rate of 114,5 kbps, each of the other users can receive data from layers 1, 2, 3 at the data rate of 148,5 kbps. For this case, by reducing QoE by 8%, PF algorithm makes it possible to receive 31% increase in allocation fairness.

Table 3

Optimal resource allocation results with proportional-fair QoE for different symbol rates and amounts of video layers

Video symbol rate (ksym/s)	Video layer	Coding rate	MC type	Symbol rate	QoE	Users, %	Average QoE	Fairness level
120	3	34	3	17	0,59	68	0,57	0,88
	2	34,5	2	23	0,42	14		
	1	80	1	80	0,23	18		
200	3	28	3	14	0,88	68	0,82	0,96
	2	36	2	24	0,80	14		
	1	162	1	162	0,66	18		
280	3	22	3	11	0,98	68	0,96	0,99
	2	25,5	2	17	0,96	14		
	1	252	1	252	0,93	18		

To analyze the possible compromises between QoE and allocation fairness, let us consider three different cases of the channel condition between the transmitter and receiver, ideal channel conditions ($C_1 = 6, C_2 = 2$).

The results depicted in Fig. 5 show that PF maximization strategy is more effective than QoE maximization to optimize the resources allocation in terms of QoE and allocation fairness, because it greatly improves fairness with a slight decrease in QoE.

In the harsh conditions of noise in the channel, QoE is growing very slowly with increase in rate when the value of the rate is below a certain threshold or higher than another threshold, as shown in the

S-curve in Fig. 4. Within this range, QoE which depend on the video data rate are slightly different, so the resource allocation strategy slightly affects QoE.

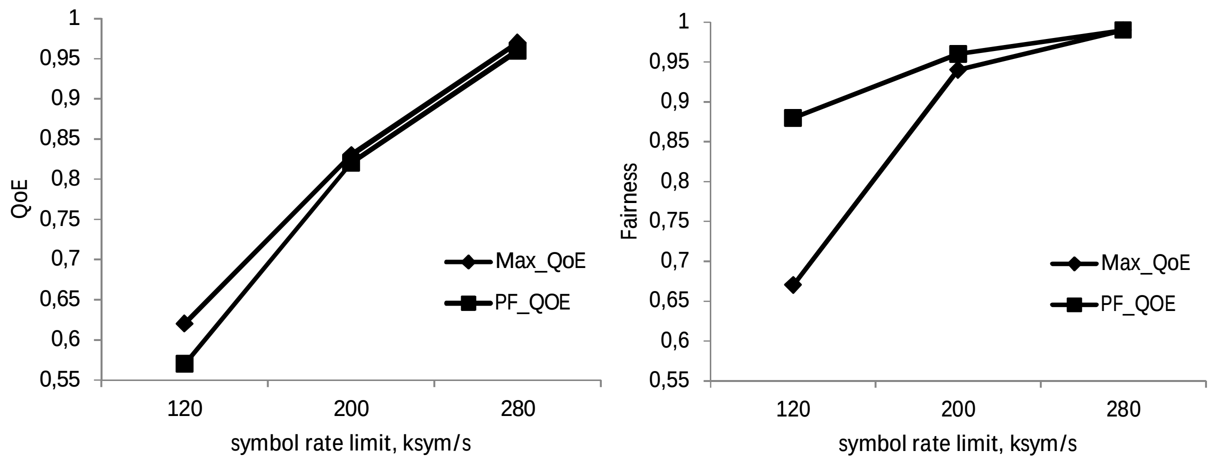

Fig. 5. Compromise between QoE and fairness of resource allocation (for $C_1 = 6, C_2 = 2$ *)*

5. Conclusions

This paper considers a problem of effective resource allocation in case of video streaming over LTE RAN infrastructure. We were analyzing the means of service quality assurance located in user and control pane of LTE and are based on LTE service architecture. We have developed the multi-layer model of group video streaming that is our simulation to calculate the objective user experience and index of proportional fairness of resource allocation. In the other hand, we have used an AMC to guarantee the proper SNR in the simulated LTE cell. The simulation was performed using the uniform distribution of subscribers around the cell in different conditions of noise in the LTE channel.

In case of bandwidth limitation to 120 ksym/s, QoE maximization strategy forms a single layer video stream and covers 68 of users only. And we are using 3d MC level. On the other hand, PF algorithm divides the resources into 3 layers and covers 100% of users by adapting MC of the lower levels to layers 1, 2 and 3 to get more coverage. Therefore, each of 18% users receives video from layer 1 at 80 kbps data rate, each of 14% users receives data from both layers 1 and 2 at an aggregate data rate of 114,5 kbps, each of the other users can receive data from layers 1, 2, 3 at the data rate of 148,5 kbps. For this case, by reducing QoE by 8%, PF algorithm makes it possible to receive 31% increase in allocation fairness.

In the harsh conditions of noise in the channel, QoE is growing very slowly with increase in rate when the value of the rate is below a certain threshold or higher than another threshold (see Fig. 4). Within this range, QoE which depend on the video data rate are slightly different, so the resource allocation strategy slightly affects QoE.

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