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A NOVEL METHOD FOR THE CODECS' PERFORMANCE ANALYSIS IN MOBILE TELEPHONY SYSTEMS

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Abstract. *This paper presents a novel method of expressing the quality of service in a mobile telecommunication system when its performance depends on several factors including applied codecs' characteristics (voice quality and data flow rate) and telecommunications traffic service possibilities. The influence of these factors is unified in one variable - quality of service measure. The proposed method is especially applicable in the cases when two-dimensional systems are analyzed – for example when two codecs with different flow rate and different achievable connection quality are used in a system. As an example, we also studied system with full-rate or mixed full-rate and half-rate codec implementation depending on the offered traffic. The system performances – mean data-flow and mean connection quality as a function of offered traffic are presented graphically and also expressed quantitatively by the novel quality of service measure. The systems with different number of available traffic channels may be compared on the base of this novel evaluation value such that the system with the highest value is the most suitable one for the concrete situation. In this way mobile system design is simplified to the great extent. The developed model is applicable generally for mobile telephony systems defining, but in this paper we studied its implementation for Global System for Mobile communications.*

Key words: *Quality of service measure, full-rate codec, half-rate codec, E-model*

1. INTRODUCTION

Mobile telephony systems are characterized by a number of parameters. These parameters include, but are not limited to base station emission power, offered and served traffic, and achieved data-flow rate. Different codec types are applied to satisfy traffic demands, each one with its specific voice quality level. When these characteristics are analyzed, usually one of them is calculated or simulated. The behaviour of the others is then only qualitatively estimated without putting it in relation to the analyzed one.

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That is why it is then difficult to estimate whether the improvement of one characteristic is overcome by the degradation in some other parameter.

There is not a significant number of contributions, which emphasizes the aggregate influence of two or more factors on some system characteristic by some unified measure (unit). One such an estimation may be found in [1], where classical equipment impairment factor I_e of the coder, defined in [2], is modified according to the influence of necessary codec bandwidth and processing time, forming the unified estimation called modified impairment factor I_b. Instead of analyzing common influence of several elements important for connection quality, the majority of contributions are directed towards improving original voice quality model [3], [4], or comparing of practically obtained characteristics for single Voice over Internet Protocol (VoIP) and mobile telephony codecs to theoretical characteristics [5]. The various testing approaches when considering only one voice connection quality are emphasized in [6]. The analysis may be performed on a per voice sample basis or on a per call basis when it is important to include the effect of recency and the effect of a speech sample with the worst signal quality. The statistical tests implementable for voice quality determination are also analysed in [6]. In some cases it is important to spread the analysis results of voice and data transmission in such a sense that they are representative for different networks (Global System for Mobile communications (GSM), Universal Mobile Telecommunication Systems (UMTS), Code Division Multiple Access (CDMA)) or application classes (emergency, business and personal). The methodology, which includes aggregate collected data on the base of user area distribution simulated by drive test and the obtained results are shown in [7]. The methodology may be also applied to compare various network providers in some area on the base of criteria how they satisfy a set of analyzed Key Performance Indicators (KPI) [8]. Very often used approach for presenting influence of two factors on the third variable is over three-dimensional (3D) graphics. Again, when considering voice coders implementation, an example may be found in [9]. But, such graphics, although are illustrative, suffer from the lack that they are often less clear for values determination than in the case of two-dimensional (2D) graphics and that it is impossible to present the influence of more than two factors on the analyzed variable. The unified numerical measure according to the results from [1], or from this paper is simple for the estimation whether the analyzed characteristic has the satisfactory value.

When projecting a telephony system such as mobile telephony system, there are several elements for the consideration to determine its quality of service (QoS). The first element is the offered traffic, which may be served with the pre-defined loss probability. The second element is the achieved mean voice connection quality, as well the quality of each, separate realized connection. The third element is necessary bandwidth to serve the offered traffic. The available frequency spectrum is very limited in mobile telephony comparing to the necessary channel capacity. That is why a number of low bit-rate codecs are developed for the implementation in mobile telephony. However, these codecs have lower voice connection quality then higher bit-rate codecs (in mobile telephony with applied GSM technology full-rate (FR – GSM 06.10) is one of them). The service principle is to use higher bit-rate codecs in all connections at lower traffic loads (usually FR in systems of the second generation (2G) or GSM). If the offered traffic overcomes the predefined level and if it is possible that requested traffic loss is not achieved, it is necessary to start the implementation of lower bit-rate codecs (usually half-rate (HR -

GSM 06.20) codec in GSM systems) in all or in a part of realized connections. Besides the limitation in available frequency spectrum, the second element of restriction is the number of dedicated traffic channels, i.e. the system hardware performance. In mobile telephony, there is possibility to even two times increase system channel capacity by the implementation of HR codec. In this case, each traffic channel may contain two connections coded by half-rate codec. A system with emphasized characteristics is analyzed in [10].

There is a variety of different parameters which may be analysed together to obtain different types of system performance estimations. The reference [11] is a comprehensive survey of elements characterizing each fifth-generation (5G) mobile system. Various among these elements besides those specific for the quality of applied codecs may be combined to obtain a number of different evaluations and [11] gives just an idea about the set of combinational factors. The most important elements which may be combined are energy consumption, transmission delay, data throughput and the applied frequency spectrum bandwidth. Generally speaking, the 5G wireless technology is based upon modified fourth generation (4G), which at present is facing many problems to meet its performance goals. The 5G wireless technology helps to solve the problems of poor coverage, bad interconnectivity, poor quality of service and flexibility [12]. 5G systems bring significantly increased data transfer speed, spectrum bandwidth, spectral efficiency and so on, comparing to 4G systems [13].

This paper describes a novel method for comprehensive analysis of different codecs applied in mobile telephony systems. It unifies different codecs' characteristics (such as their achievable voice quality, data throughput and possibilities for traffic serving) into one variable, which may be compared for various codecs. The main goal of the research is to simplify mobile system designing (selection of the number of traffic channels) to achieve the best system performance while considering several KPIs. In the paper, the analysis is, first of all, limited to codecs implemented in GSM systems. The number of traffic channels for which the results are presented is specific for GSM systems with 1, 2, 3 or 4 frequency carriers. Such analysis, unifying different mobile telephony codecs' characteristics into one variable are rarely implemented and they, if applied, unify together lower number of codecs' characteristics.

The section 2 in the paper describes E-model, which we implemented for voice quality estimation as the first element included in our unified variable. The values of data-flow rate as the second element of this evaluation are also cited at the end of this section. The traffic model corresponding to the GSM mobile telephony system with two different applied codecs is developed in the section 3 as the third element in the analysis. Section 4 describes how the value of our novel estimation is determined and presents the obtained results. Section 5 is a brief survey how the model could be implemented in mobile systems of the third generation (3G) and the fourth generation (4G). At the end, section 6 is the conclusion of the paper.

2. Voice Connection Quality Estimation

There are several subjective and objective models intended to express the voice connection quality. Analysis in this paper is based on E-model implementation. E-model is the computational, objective model. It joins the influence of various factors into one unique quality measure – rating factor R [2]. The value of R is connected with Mean Opinion Score (MOS), which is between 1 and 5. The values of MOS and R are connected by formula and corresponding Fig. B.2 from [2].

The main purpose of E-model is to express voice connection quality in a number of different kinds of voice connections systems. Among them implementation in Internet packet connections quality estimation is, probably, the most often one. According to [14], MOS is used as the measure of voice quality in mobile telephone connections. Taking into account that MOS and R are mutually dependent variables, E-model may be implemented as a measure for the estimation of the quality of mobile telephony connection.

According to E-model, connection-rating factor is [2]:

$$R = 94 - I_{e-eff} \tag{1}$$

where $R0$=94 is the maximum practically possible connection-rating factor. Effective equipment impairment factor $I_{e\text{-}eff}$ includes impairments caused by the codec implementation and influence of random signal loss. In (1) we do not consider influence of factors which appear in original equation from [2]: combination of impairments which occur simultaneously with voice signal (I_s), influence of delay (I_d) and influence of advantage factor (A), because these factors are not related to the implemented codec.

The value of $I_{e\text{-}eff}$ is calculated from the equation [2]:

$$I_{e-eff} = I_e + (95 - I_e) \cdot \frac{P_{pl}}{\frac{P_{pl}}{BurstR} + B_{pl}} \tag{2}$$

where it is:

- I_e – equipment impairment factor when there is no packet loss;
- P_{pl} – transmitted signal loss probability (in percent);
- *BurstR* – burst ratio: the quotient of the average lengths of the lost signal parts in real transmitted signal and when signal parts are randomly lost;
- B_{pl} – robustness factor, which is specific for each coder type.

In this paper, we are limited to cases when there is no signal loss (P_{pl}=0). Such a situation has not the significant probability in mobile telephony systems, but the obtained results do not suffer from the loss of generality. It means that it is $I_{e\text{-}eff}=I_e$ from (2), i.e. $R=94-I_e$ from (1) in our analysis. According to the available literature, we have chosen the values I_{eFR}=20 for FR codec and I_{eHR}=23 for HR codec [15]. However, also the worse, pessimistic value I_{eFR}=26 may be found in some references [16], [17]. The reason is that the value of R should be higher for FR codec than for HR codec, because data-flow is greater in the case of FR codec. Besides, it is explained in [5] that, according to measurements, even the value I_{eHR}=23 is pessimistic for HR codec.

Data-flow rate for FR and HR is also important in our analysis. It is DF_{FR}=13kbit/s for FR codec and DF_{HR}=5.6kbit/s for HR codec. Besides this, in GSM systems enhanced full-rate codec (EFR) may be also applied. Its data-flow is DF_{EFR}=12.2kbit/s.

3. Implemented Traffic Model

Traffic model of the analyzed system is presented in Fig. 1. The total number of traffic channels is N and each channel, if busy, may contain one FR or one or two HR connections. Each system state is modelled by two-dimensional variable $\{n_f,n_h\}$, where n_f and n_h are the numbers of instantaneously realized FR (i.e. HR) connections, respectively. The threshold number of channels when HR connections begin to be established is K ($\leq N$). As a consequence, the total number of available traffic channels is equal

$$N_c = K + 2 \cdot (N - K) \tag{3}$$

The value of K is chosen on the base of the request that traffic loss is lower than some limiting value, in our analysis it is P_{loss}=1% or P_{loss}=2%. The intensity of new requests generation is λ and the probability that the user, who generates the request, may establish half-rate connection, is π_h. In up-to-date technology it may be supposed that the majority of mobile stations have the possibility to realize a HR connection. So we suppose that the value of π_h is in the range between 0.8 and 1. Call duration of both connection types is random variable with exponential distribution and mean duration $t_p=1/\mu$. In the period when both kinds of requests are generated, the intensity of full-rate requests generation is $\lambda\cdot(1-\pi_h)$, while the intensity of half-rate requests generation is $\lambda\cdot\pi_h$. In the state $\{n_f,n_h\}$ full-rate connection is finished with the intensity $n_f\cdot\mu$ and half-rate connection with the intensity $n_h\cdot\mu$. If during traffic process realization happens that two one-half time slots appear (i.e. two channels with only one half-rate call per channel), these two calls are gathered in one completely busy channel, while the other channel becomes completely idle (complete re-packing, [18]).

Figure 1 presents the model of a system with HR connections realization possibility. The number of traffic channels is N and the threshold when HR connections realization starts is K=3. The value of π_h is $0<\pi_h<1$. This is a two-dimensional Markov birth-death traffic model. The solution may not be obtained in the closed form, because Kolmogorov's criterion for system reversibility is not satisfied [19]. Kolmogorov's criterion is satisfied in the special case of high traffic load when it is necessary to have K=0 to achieve satisfactory traffic loss. The system remains two-dimensional and it is possible to realize circular flow between any set of four system states in both clockwise and counter clockwise direction. For a system in Fig. 1 such condition is not satisfied for states where there are less than 3 FR calls because there is no circular flow in counter clockwise direction.

There are two more special cases when the system is simplified for the analysis comparing to the system from Fig. 1. The first one is for a low traffic load when there is no need to implement HR connections, but only FR. In this case, we obtain one-dimensional traffic model with N+1 system states (the states are {0,0}, {1,0}, ..., {N,0}, i.e. it is always n_h=0). The second special case is for the higher traffic load where it is K=0, while in the same time it is π_h=1. Under these conditions we also obtain one-dimensional traffic model with $2\cdot N$+1 system states where only HR connections are realized (the states are {0,0}, {0,1}, ..., {0,$2\cdot N$ }, i.e. it is always n_f=0).

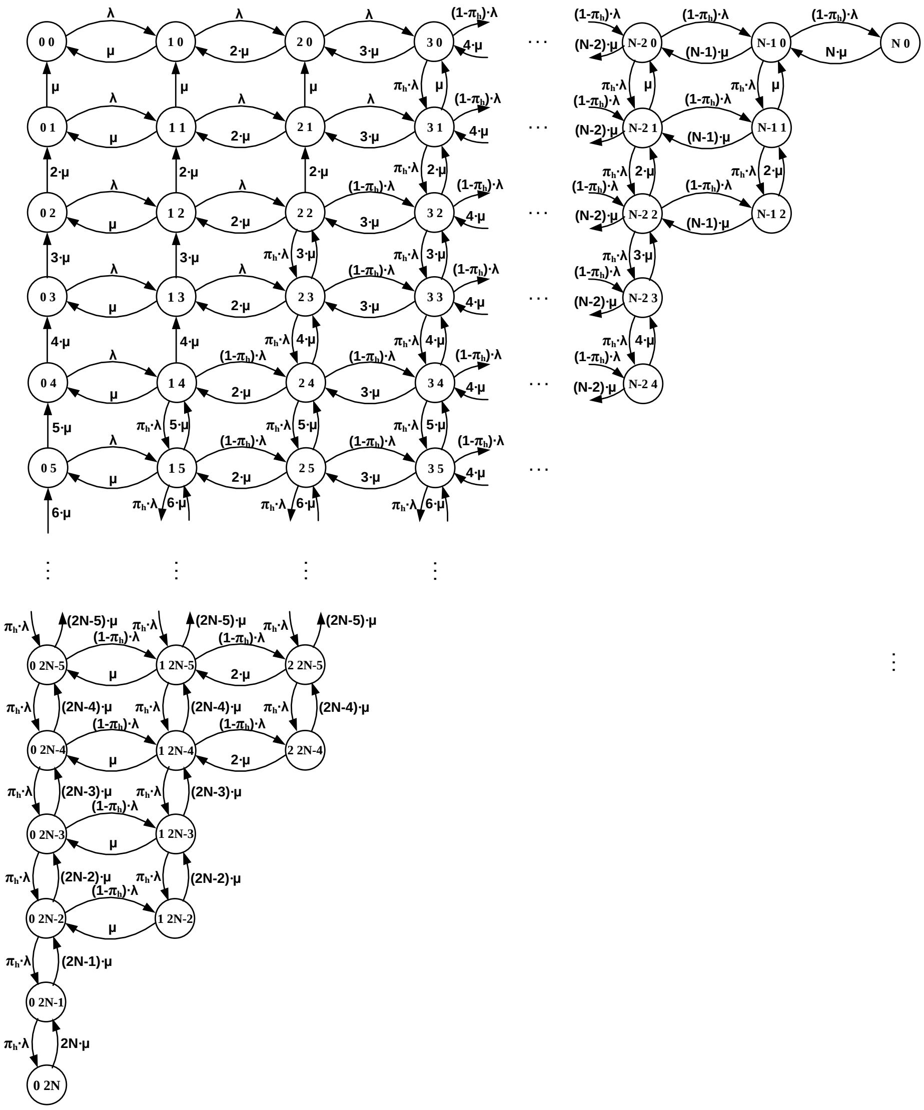

Fig. 1 Birth-death model of a system with FR and HR connection realization possibility

In the general case presented in Fig. 1 the solution is obtained by solving the system of equations [10]. The equations in stationary case are obtained by equating the transition intensity from some system state $\{n_f,n_h\}$ with the total transition intensity into the same state. This is expressed by the equation

$$\begin{aligned}P(n_f,n_h)\cdot(\lambda(n_f,n_h)+\mu(n_f,n_h)) &= P(n_{f-1},n_h)\cdot\lambda(n_{f-1},n_h)+P(n_f,n_{h-1})\cdot\lambda(n_f,n_{h-1})+\\ &+P(n_{f+1},n_h)\cdot\mu(n_{f+1},n_h)+P(n_f,n_{h+1})\cdot\mu(n_f,n_{h+1})\end{aligned} \tag{4}$$

where $P(x,y)$ is the probability of state $\{x,y\}$, $\lambda(x,y)$ is the intensity of new call generation in the state $\{x,y\}$ and $\mu(x,y)$ is the intensity of call termination in the same state $\{x,y\}$. In order to successfully solve the system of equations, it is necessary to insert also the condition

$$\sum_{n_f=0}^{n_f+2\cdot n_h=N}\sum_{n_h=0}P_{n_f,n_h}=1 \tag{5}$$

instead of one of the equations expressed by (4).

Besides of solving the system of equations, the system from Fig. 1 may be also analyzed by simulation program. Such a program is our original development realized in C programming language and it is executed on commercial personal computer. The program is the improved version of the simulation program, which is described and verified in [10] by comparing the values of state probabilities to the values obtained by solving the system of equations for a system with relatively low number of channels.

4. Method of Analysis and the Results

The main condition to perform the analysis presented in this paper is to define the connection scenario which allows optimum relation among three variables: 1. offered traffic; 2. traffic loss and 3. implementation as low as possible percent of HR connections (i.e. connection quality).

The first step in the analysis is to determine components of served traffic (total traffic A_s, traffic of FR connections (A_{sf}), traffic of HR connections (A_{sh}), as well as the loss probability of these three traffic components (P_{loss}, P_{lossf}, P_{lossh}) for each value of predefined traffic, while changing the threshold number of channels K from 0 to N. These values of served traffic and traffic loss are obtained in the simulation process. In our analysis we predefined the probability π_h=0.8. At the end of simulation the components of offered traffic (total offered traffic - A_o, offered traffic of FR connections - A_{of} and offered traffic of HR connections - A_{oh}) are calculated as:

$$A_o=\frac{A_s}{1-P_{loss}} \tag{6}$$

$$A_{of}=\frac{A_{sf}}{1-P_{lossf}} \tag{7}$$

$$A_{oh}=\frac{A_{sh}}{1-P_{lossh}} \tag{8}$$

The goal of this analysis is to determine the maximum values A_s, A_{sf} and A_{sh} for which is P_{loss} lower than the predefined value (1% or 2% in our simulation) and the corresponding value of K.

The second step in the analysis is calculation of mean rating factor R value for A_{sf} and A_{sh} determined in step 1. This value is

$$R_{mean} = \frac{(94 - I_{eFR}) \cdot A_{sf} + (94 - I_{eHR}) \cdot A_{sh}}{A_{sf} + A_{sh}} \tag{9}$$

The third step is the calculation of mean data-flow value for the same values of A_{sf} and A_{sh} as in step 2. The implemented formula is

$$DF_{mean} = \frac{DF_{FR} \cdot A_{sf} + DF_{HR} \cdot A_{sh}}{A_{sf} + A_{sh}} \tag{10}$$

And finally the fourth step is calculation of our novel quality of service measure ($QoSM$). Its value is

$$QoSM = \frac{A_o \cdot R_{mean}}{DF_{mean} \cdot N} \cdot 100 \tag{11}$$

Such an expression is formed, because a system has better characteristics when it may serve higher traffic with higher achieved voice quality, but with lower data flow. The division by the number of traffic channels is introduced in order to be able to compare performances of the systems with different number of traffic channels, meaning that the unit of this variable is expressed per one channel. This novel variable contains two factors related to the performances of applied codec (data flow rate and connection-rating factor). The third component is related to more general element, which is specified for each telecommunication system independent of the implemented codec type – offered, i.e. served traffic. In this way the defined variable is not just related to codec properties, but is more comprehensive, describing service in a system.

Table 1 presents the value of threshold number of traffic channels (K) when the implementation of HR codec starts. The value K is determined as a function of the value of offered traffic (A_o), which is defined at the beginning of simulation for the systems with 6, 14, 22 and 30 traffic channels (N). These values of N are characteristic for the systems with 1, 2, 3 and 4 frequency carriers, respectively. The values of K are presented for offered traffic loss P_{loss}=1% and P_{loss}=2%. The designations P_{loss}>1% and P_{loss}>2% in the Table 1 mean that the predefined value of offered traffic loss may not be achieved regardless of K.

As an example, let us consider the system with A_o=17E and desired P_{loss}=2%. According to Table 1, it is not possible to achieve P_{loss}=2% by systems with 6 and 14 traffic channels. When there are 22 traffic channels, it is necessary to have mixed FR and HR connections and the threshold number of channels when HR connections are started to be realized is K=20. In the systems with N=30 traffic channels all connections may be FR (K=30).

Table 1 The threshold number of channels (K) as a function of predefined offered traffic (A) and total number of channels (N)

$A_o(E)$	P_{loss} = 1%				P_{loss} = 2%			
	N				N			
	6	14	22	30	6	14	22	30
	K				K			
1.5	6	14	22	30	6	14	22	30
2	5	14	22	30	6	14	22	30
2.5	4	14	22	30	5	14	22	30
3	3	14	22	30	4	14	22	30
3.5	2	14	22	30	4	14	22	30
4	1	14	22	30	3	14	22	30
4.5		14	22	30	2	14	22	30
5		14	22	30		14	22	30
6		14	22	30		14	22	30
7		14	22	30		14	22	30
8		13	22	30		14	22	30
9		12	22	30		13	22	30
10		11	22	30		12	22	30
11		10	22	30		11	22	30
12		9	22	30		11	22	30
13		8	22	30		10	22	30
14		6	21	30		9	22	30
15			20	30		7	21	30
16			20	30			21	30
17			19	30			20	30
18			19	30			20	30
19			18	30			19	30
20			18	30			19	30
21	P_{loss}>1%		17	29			18	30
22			16	28	P_{loss}>2%		18	29
23			15	28			17	29
24			14	27			16	28
25			12	27			16	28
26				26			15	27
27		P_{loss}>1%		26			13	27
28				25		P_{loss}>2%		27
29				25				26
30				25				26
31				24				26
32			P_{loss}>1%	24				25
33				23			P_{loss}>2%	25
34				22				24
35				21				24
36				20				23
37				17				23
38				P_{loss}>1				22
39				%				20

Figure 2 presents relation between A_o, DF_{mean} and R_{mean} for a system with 6 traffic channels when traffic loss is limited to 2%. The threshold number of channels is determined according to data in Table 1. The values of DF_{mean} and R_{mean} are determined on the base of expressions (10) and (9), respectively, based on A_{sf} and A_{sh} obtained in simulation process. After that, the values of $QoSM$ are presented in Fig. 3 according to (11) for the same values of A_o, DF_{mean} and R_{mean} as the ones used for the graphic in Fig. 2. The results are presented both for traffic loss less than 1% and less than 2%.

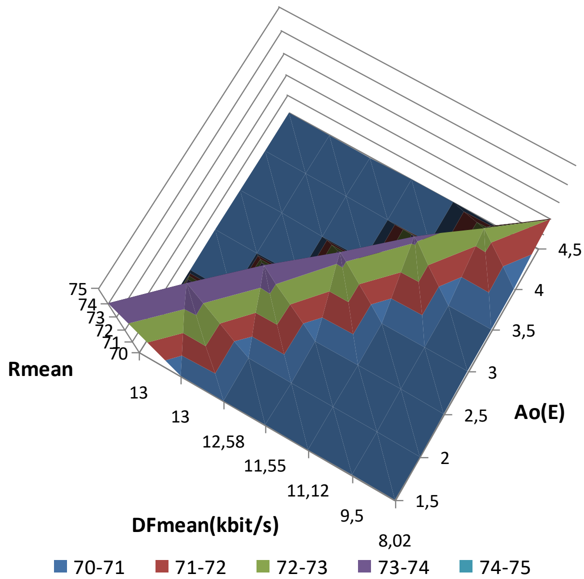

Fig. 2 Mean connection-rating factor R_{mean} as a function of offered traffic A_o and mean data-flow DF_{mean} for a system with 6 traffic channels and probability of traffic loss P_{loss}<2%

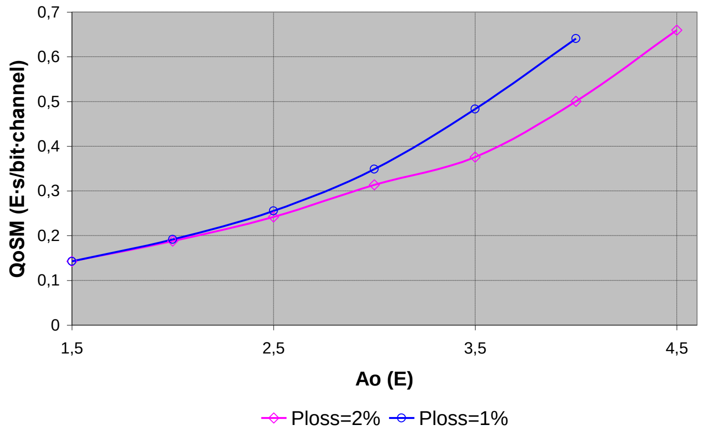

Fig. 3 $QoSM$ as a function of offered traffic A_o for a system with 6 traffic channels

Figure 4 and Fig. 5 present the same variables as the Fig. 2 and Fig. 3, but for 14 traffic channels. After that, the similar analysis results are presented in Fig. 6 and Fig. 7 for the system with 22 traffic channels and in Fig. 8 and Fig. 9 for the system with 30 traffic channels.

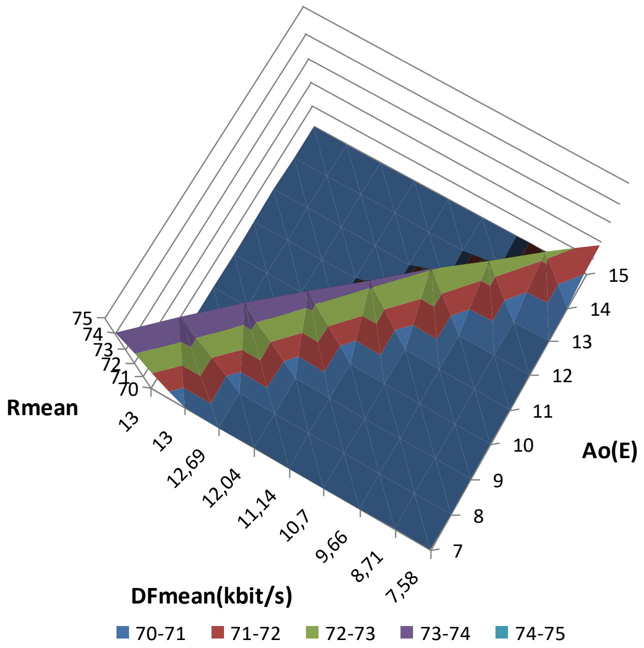

Fig. 4 Mean connection-rating factor R_{mean} as a function of offered traffic A_o and mean data-flow DF_{mean} for a system with 14 traffic channels and probability of traffic loss P_{loss}<2%

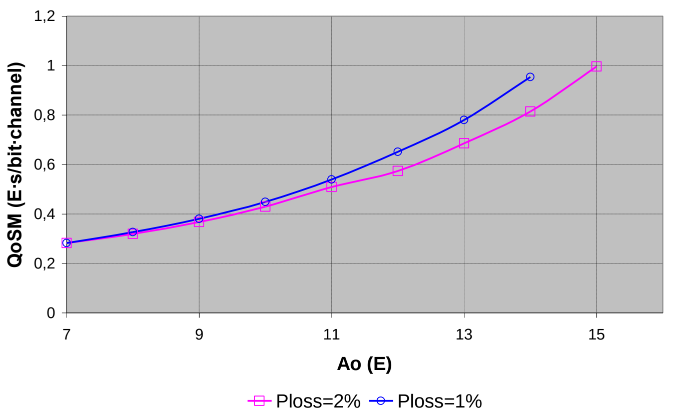

Fig. 5 *QoSM* as a function of offered traffic A_o for a system with 14 traffic channels

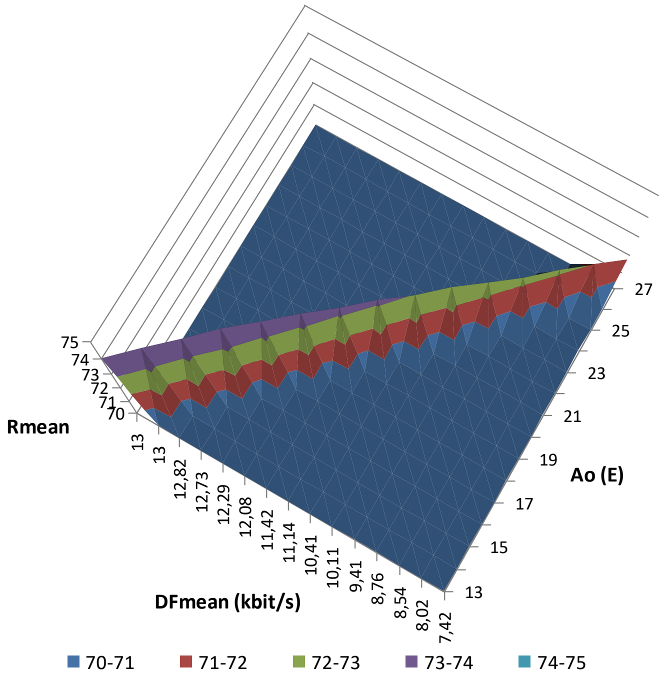

Fig. 6 Mean connection-rating factor R_{mean} as a function of offered traffic A_o and mean data-flow DF_{mean} for a system with 22 traffic channels and probability of traffic loss P_{loss}<2%

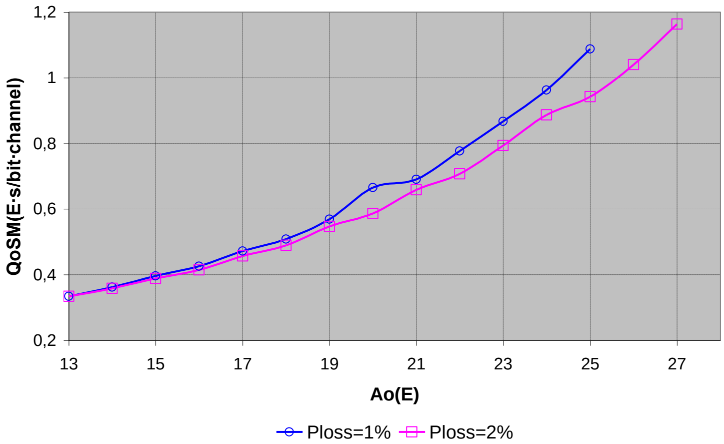

Fig. 7 *QoSM* as a function of offered traffic A_o for a system with 22 traffic channels

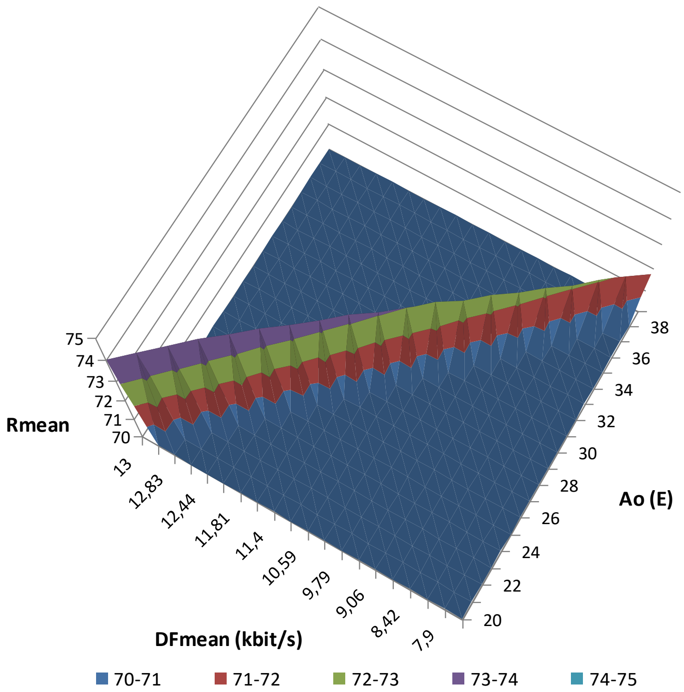

Fig. 8 Mean connection-rating factor R_{mean} as a function of offered traffic A_o and mean data-flow DF_{mean} for a system with 30 traffic channels and probability of traffic loss P_{loss}<2%

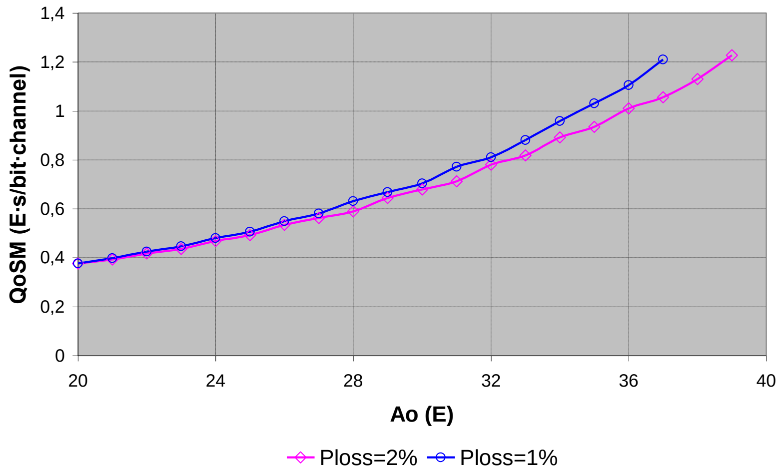

Fig. 9 *QoSM* as a function of offered traffic A_o for a system with 30 traffic channels

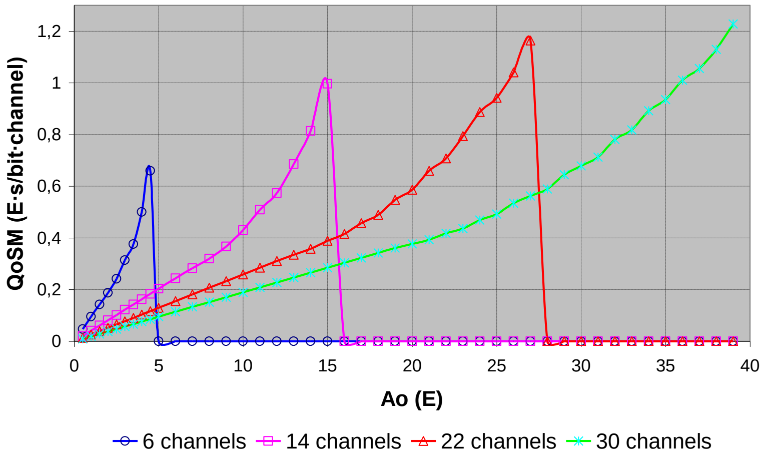

Fig. 10 *QoSM* as a function of offered traffic A_o for systems with 6, 14, 22 and 30 traffic channels and probability of traffic loss P_{loss}<2% supposing that *QoSM*=0 if P_{loss}=2% may not be achieved

Figure 10 illustrates the comparative results of *QoSM* for systems with 6, 14, 22 and 30 channels when it is P_{loss}<2%. The situation when it is P_{loss}>2% is treated as that specified characteristics are not satisfied and that's why in that case the value of Q_oSM suddenly drops to 0. The system dimension (i.e. the number of traffic channels) is selected to achieve the highest value of *QoSM*. For example, if it is A_o=6E, we have to choose system with 10 channels, because it has higher *QoSM* than other 3 systems.

5. Model Implementation in 3G and 4G Systems

The model, which is developed in this paper, is also applicable for the analysis of 3G and 4G systems. The calculation procedure is here presented for the GSM (2G) systems and its FR and HR codec. In order to discuss model applicability in 3G and 4G systems, it is necessary first to perform a brief survey of codecs, which are implemented in 3G and 4G systems.

Adaptive multirate (AMR) codec is usually applied in 3G and 4G systems when standard telephony frequency bandwidth 300Hz-3400Hz is applied [20]. Here we are focused on AMR codec, as it is comparable to FR and HR codec, which we considered in previous sections. At this moment we do not include AMR WB (adaptive multi-rate wideband), EVRC (enhanced variable rate codec) and other codec types intended for wideband signal coding (till 7kHz or more), as well as other narrow-band signal codecs, which are used less than AMR (for example, iLBC (internet low bit-rate codec).

AMR codec is based on the implementation of FR and HR codec, but with adaptable voice bit-rate (codec mode) according to signal transmission conditions [20]. There are 8 different codec modes and their bit-rates are between 4.75kbit/s and 12.2kbit/s. After selecting corresponding codec mode, the coded signal is transmitted in FR or HR channel (channel mode). It may be said that in 3G systems voice signal is transmitted "traditionally", as also in GSM systems. When codec bit-rate is decreased, more bits are spared for error protection. On the base of this consideration, it is possible to conclude that equation (10) remains valid also when AMR codec in 3G systems is considered with the same values of data-flow, as FR and HR channels are steel implemented. Equation (9) is also valid, but the concrete values of I_e are changed and they depend on the voice bit-rate. These values are between 19 when voice bit-rate is 4.75kb/s and 3 when voice bit-rate is 12.2kb/s [21], as calculated on the base of the presented connection-rating factor. They are improved comparing to already emphasized values for implementation in GSM systems. The values of I_e for AMR codec are not always explicitly defined. For example, in [22] these values are different in various performed tests and the best results are approaching the values from [21]. The values of I_e in [22] may be calculated on the base of the presented values of Mean Opinion Score (MOS).

The part of our analysis related to traffic demands in model structure does not depend on codec modifications. There are still two channels' types and equations (6)-(8) are valid. The value of π_h additionally depends on signal transmission conditions, as these conditions have influence on the choice of corresponding voice bit-rate. At the end, the main equation (11) may be also applied for *QoSM* factor calculation.

The technology of signal transmission is a bit different in the case of 4G systems. The applied codecs are dominantly the same as in 3G systems [23], meaning that for our analysis AMR codec is important. However, the applied frame structures are different, as they are based on VoIP technology [24]. More precisely, it is voice over long term evolution (VoLTE) [25]. As a consequence, we may say that the part of our model related to voice quality, i.e. equation (9), remains as for 3G systems. The considerations dealing with data-flow rate (equation (10)) overcome the scope of this paper and will be studied in the future. Overall, the detailed analysis of presented model in 3G and 4G systems will be the subject of our future development. Besides analysis for AMR codec, this analysis should also incorporate adaptive multirate wideband (AMR WB) codec, as it is emphasized in [25] that its implementation is mandatory in VoLTE systems.

6. Conclusion

In this paper, we analyzed the performances of systems with mixed traffic realization where two different traffic components are defined by the implementation of two codec types. The components of traffic (i.e. two applied codecs) differ in the connection quality. Until the traffic threshold, which is determined according to the allowed traffic loss, only better quality codec is implemented for connection realization. After that for the higher traffic two codec types are applied. A primer of such a system in mobile telephony, which is analyzed in this paper, is a system where full rate (FR) codec has a better quality and half rate codec (HR) has a lower quality. The three main quality elements of such a system: offered traffic (A_o), mean connection quality (R_{mean}) and mean data-flow (DF_{mean}) are mutually dependent and their relation is demonstrated by 3D graph. After this, first contribution, the second more important contribution is the definition of a novel *QoSM*, which allows us to express cumulative influence of three previously cited system performances by one value. This is a novel approach, which may be also implemented in the other situations when it is necessary to analyze common influence of several factors. The scope of the analysis is to facilitate mobile systems design and mutual comparison.

The variable *QoSM* is defined in such a way that a number of necessary traffic channels is selected easily on the base of highest *QoSM* value from the graph. According to definition and in real physical sense, *QoSM* is increased when A_o and R_{mean} are increased, but also when DF_{mean} is decreased. On the base of graphs from this paper, the benefits expressed by better system utilization when the offered and served traffic are increased while in the same time necessary data-flow is decreased significantly overcome effects of simultaneous connection quality degradation.

There are two directions of our future activities. The first one is connected with already performed development. The model presented in this paper is universal for the implementation in mobile telephony. In our analysis we are limited to GSM systems. We proved by a brief survey the method applicability for the implementation in 3G and 4G systems. Model adaptation for such an application will be the subject of future development.

The second direction of our activities is directed towards model further improvement by involving new elements in the model. According to the considerations in the introductory section of this paper, the important elements may include energy consumption, transmission delay and the applied frequency spectrum bandwidth.

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