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# Estimation of QoS Dynamics in the Wireless Networks

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## Abstract

An important performance aspect of widespread networks is the quality of perceived conversational speech of voice calls and data services. Today these networks belong to 2-2.5 generation. Third generation (3G) networks were created only in the last decade. Currently 3G technologies are standardized. Equipment manufactures and network operators are in the process of deployment of 3G networks. Scientific institutions pay attention to 4G and generally to next-generation telecommunication networks. The QoS for individual network user is becoming the main question of nowadays research.

In this paper we investigate the possibilities and means to create special purpose module for the evaluation of perceived service quality. It is presented that demand for such module can be justified since the quality perceived by individual users varies greatly. The paper also shows that QoS depends on user location, time, network load etc. The aim of the presented work is to find better relations between ISP and end-user from QoS point of view.

## Keywords

iQoS, Wireless, PESQ, GSM

## Working Group 2

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In this paper we investigate possibilities and means to create special purpose module for the evaluation of perceived service quality. It is presented that demand for such module can be justified since the quality perceived by individual users varies greatly. The paper also shows that QoS depends on user location, time, network load etc. The aim of the presented work is to find better relations between ISP and end-user from QoS point of view.

## 1. Introduction

It is certain that next-generation networks will convey all information wrapped in packets. For example, voice is divided into length *T* time intervals (typically 10, 20, 30 ms) before transmitting. These intervals are encoded and transmitted. Transmitted block is named as frame. At the receiver end, it is possible that frame is damaged. Damaged frames are not normally decoded and decoder uses some substitution algorithm.

According to ITU – QoS is a conditional subject. However, the main criterion is "the satisfaction of the end user". This formulation of QoS permits quality evaluation at the application level. The degree of satisfaction depends on the perceived service level, user expectations and cost. Service level is merely a reflection of the personal impression of the users, and consequently the best way to estimate its testing conducted with a selected set of persons.

One such demand is in finding means to evaluate QoS provided de facto to the individual user – iQoS [1-4].

In this paper we will investigate possibilities and means to create special purpose module for the evaluation of *de facto* perceived service quality. Demand for such module can be justified since the quality perceived by individual users varies greatly. Quality depends on user location, time, network load etc. When user moves the communication conditions changes as well as perceived quality level.

Attitude towards QoS from user perspective is relatively new. Traditionally quality related quantities are statistically processed. Averaging is usually done for the whole network.

## 2. *De facto* Provided QoS

The modern telecommunications networks differentiated services. Distinct services require particular quality levels or aspects. Basically, there are three kinds of service models: Best-Effort Service, Integrated Service, and Differentiated Service. For example, Differentiated Service - a multiple service model that can satisfy different QoS requirements. For the Best-Effort Service, the network delivers data if it is possible, without any assurance of reliability, delay bounds, or throughput.

Distinct service classes are well suited to supply needs of different users. Services of higher class require more network resources. It is necessary to reserve some network resources when supplying

Differentiated Services. This reservation limits potential and partially quality of lower class services. From above it is obvious that higher level services are more heavily priced.

In the mobile telecommunication networks QoS as concept and a set of some parameters is used for network resource management. Telecommunication equipment constantly monitors communication conditions in mobile and base stations. Changes of communication conditions lead to change in system parameters. The power of transmitted signal is changed first. When increased power is not enough to hold good communication conditions then it is possible to change frequency channel, mobile station can be switched to another base station, etc. Third generation systems can also change voice codec parameters. Such means give to the mobile user some invariability limits. When mobile station moves in relatively small territory, quality level changes too; these changes, however, but mostly slight.

In speech services UMTS uses AMR voice codec [7]. Voice quality depends on coding rate (Table 1, according to [13].

Table 1: AMR Codec Quality Rate According MOS[1], and MOS

| AMR Codec Mode (kb/s) | 4,75 | 5,9 | 6,7 | 7,4 | 7,95 | 10,2 | 12,2 |
|---|---|---|---|---|---|---|---|
| MOS[1] (No Errors) | 3,5 | 3,7 | 3,77 | 3,83 | 3,91 | 4,06 | 4,01 |
| MOS (C/I 7dB) | 3,52 | 3,7 | 3,96 | 3,94 | 3,96 | 3,9 | 3,44 |

In GSM/GPRS and UMTS systems radio frequency (RF) power control is employed to minimize the transmit power required by mobile station (MS) or base station (BS) while maintaining the quality of the radio links. By minimizing the transmit power levels, interference to co-channel users is reduced. RF power control is implemented in the mobile station on each uplink channel and optionally in the base station. In GSM/GPRS radio channel quality $Q$ is measured and power control is performed with the some measured/calculated parameters: received signal level (RxLev), carrier-to-interference ratio (CIR), bit-error rate (BER), frame erasure rate (FER) or received signal quality (RxQual) [6].

The power control mechanism in principle is quite simple. If $i$-th radio channel quality $Q_i$ is better than required $Q^{\text{req}}$.

$$Q_i > Q^{\text{req}}, \tag{1}$$

CIR shall be decreased. When radio channel quality is worse than required,

$$Q_i < Q^{\text{req}}, \tag{2}$$

CIR shall be increased. The decision "CIR shall be increased" should be executed until

$$P_i \le P_{\max}, \tag{3}$$

here $P_{\max}$ - maximum allowed power of transmitter. While $P_i < P_{\max}$, radio channel condition variations is slightly affect quality of service. But when power is increased and critical limit is reached

$$P_i = P_{\max}. \tag{4}$$

It is not possible to increase power more. This modified power control algorithm is illustrated in Fig. 1. This model differs from conventional models because of it is impossible to increase transmitter power when $P_{\max}$ is reached.

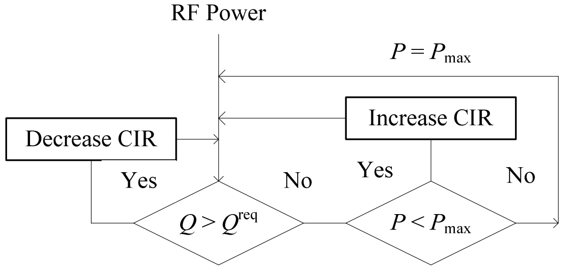


Figure 1: Power Control

When $P_{max}$ is reached and in some location it is not possible to switch MS to another BTS quality of link may become poor. Examples measurements of the parameters like RxLev and BER when mobile station moves in poor communication conditions area, are depicted in Fig. 2 and 3 [2].

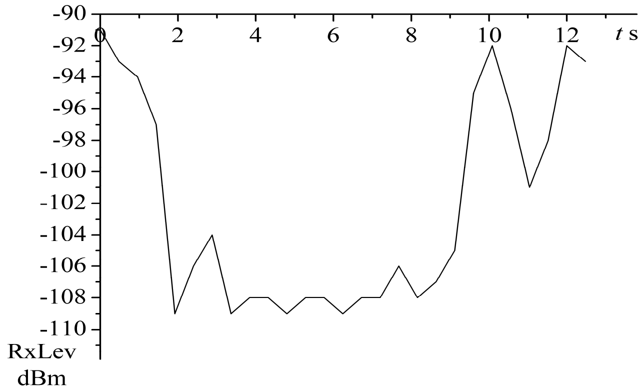


Figure 2: Received Signal Level as Function of Time (MS is moving)

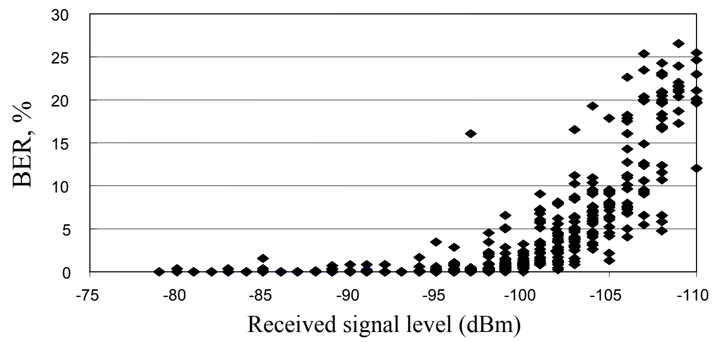


Figure 3: BER as a Function of Received Signal Level

In Fig. 2 it can be seen 8 seconds time duration interval when signal level RxLev is near to sensitivity threshold of the receiver (–110 dBm). Fig. 3 shows that BER greatly rises when received signal level falls below –95 dBm.

The measurements [2] have shown that in cases of poor radio link quality (for $-110 < RxLev < -96$ dBm), the FER > 10% was observed for 17.9% of measurement points, and C/I<6 dB was observed at 38% of all points. The occurrence of both FER > 10% and C/I < 6 dB was observed for 17% measurement points. This confirms that in the areas with poor radio coverage (radio signal strength being near to the receiver sensitivity threshold) the obtainable voice quality will be significantly degraded.

Very important question is how insufficient quality conditions are regular. The networks providers such data declare seldom. Certain exception is Ascom report [5]. This report provides the results of QVoice measurement data, recorded in Denmark. In these measurements speech quality in ≤5.3% call attempts was found fair and in≤ 2.3% call attempts was bad. Dropped calls were in ≤ 2.3% call attempts.

Ascom report also points out reasons of bad communication link: bad coverage, interference, etc. Percentage of bad quality links is not big but distribution of cases of bad communication quality is not uniform among users. Cases of bad communication quality fall to those users who are in some geographical location (usually between base stations) or use services when network load is high.

Summarizing measurements of the link quality of mobile communication we can draw conclusions that are important for iQoS problem:

- Particular MS operates under distinct conditions. These conditions depend on MS localization, network load, etc.
- When MS is moving communication conditions changes randomly, there are some time intervals when signal level approaches receiver sensitivity threshold;

- There are places in cells where elementary quality defects may occur and frames may be lost;
- When MS operates under low signal level conditions series of lost frames are observed;
- Lost single and series of frames making degradation of common capacity and direct making influence to speech and video quality degradation.

## 3. Impact of lost frames on voice quality

When mobile station operates under critical radio link conditions quality degradation can be observed. In these cases user gets only best effort service. Voice quality impairments are due to lost radio frames. Consequence of this is lost speech frames. Single erased frames are heard as short disturbances. Long sequences of lost frames erase words or part of words.

Some works have been carried out on the effects of packet loss on voice quality. Cox and Perkins compared the impact of random and burst packet loss [8]. They found that for low packet loss rates a burst distribution gave a higher subjective quality than a non bursty distribution whereas for high packet loss rates the converse was true. At the lower 1 and 2 percent rates there is no significant effect of the distribution (random vs. bursty) variable, whereas with the high 5 percent rate opinion suffers more when packets are lost in bursts. Comprehensive research on lost packets influence on speech quality is given in Tiphon report. For the packet loss conditions, frames were removed from the speech samples randomly with a frequency determined by the test condition (e.g., 1% of the frames). Relation of erased frames with voice quality is described in ITU recommendations (G.113/Appendix I. 2002).

Ding and Goubran conducted a modelling in which impairment factor grows logarithmically with increasing packet loss rate or packet size [8].

Conway [10] has proposed original methodology for the research of lost packets impact. This methodology is named Framed PESQ. A basic assumption in the proposed method is that the encoded digital speech signal is transmitted in a framed format. The erased frames are signalled to the decoder of the received encoded speech signal. An exact PESQ measurement of the speech quality is provided in a known reference signal. Framed PESQ evaluates a known reference signal instead of transmitted speech. There is a doubt whether such method is suitable for iQoS.

The development of speech quality metrics relying only on GSM transmission parameters RxQual, FER has been presented in [11, 12]. The proposed speech quality measure is an empirical function which depends on RxQual, FER.

The main purpose of these works is to predict MOS scores using measured parameters such as RxQual, FER. Authors state that it is possible to predict the subjective speech quality very accurately. The graphs presented in the mentioned work are not very impressive in the sense that maximum deviation of predicted MOS scores from true MOS scores exceeds one. In [11, 12] it is indirectly shown that FER and RxQual parameters contain enough information for voice quality prediction. It is possible to construct many kinds of models which take as an input above mentioned parameters and give as an output predicted MOS score for conversation.

## 4. Simulation results

After the review of published works it is possible to make conclusion that there is not enough data to predict voice quality when communication conditions are nearly critical. Therefore we have made single word degradation research when 1, 2, 3, …, *n* frames are lost. Different lost patterns were used – from deterministic to purely random. Simulation system is depicted in Fig. 4. The main components of a system are AMR [7] coder, AMR decoder, and PESQ measurement algorithm and packet loss imitator.

AMR codec uses 20 ms lasting packets for speech transmission. Every word consists of some of these packets. Typical value is 20-40 packets per word. In our research we will simulate packet loss in every possible position in a word. After this loss we calculate PESQ score for original and damaged words. In this way we have got dependence of PESQ score on loss position for one packet loss in a word in every possible place. Also we have simulated consecutive loss of two, three and more packets starting from every possible place in a word. Graphs of these simulations are presented in Figure 5.

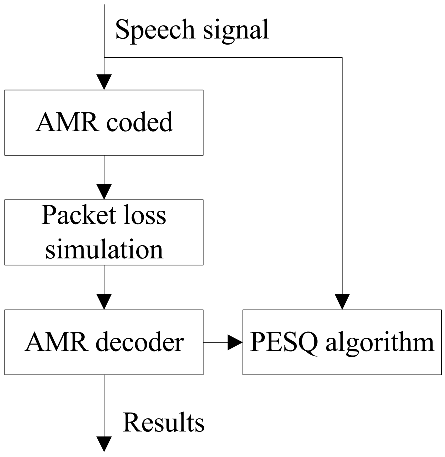


Figure 4: Simulation scheme

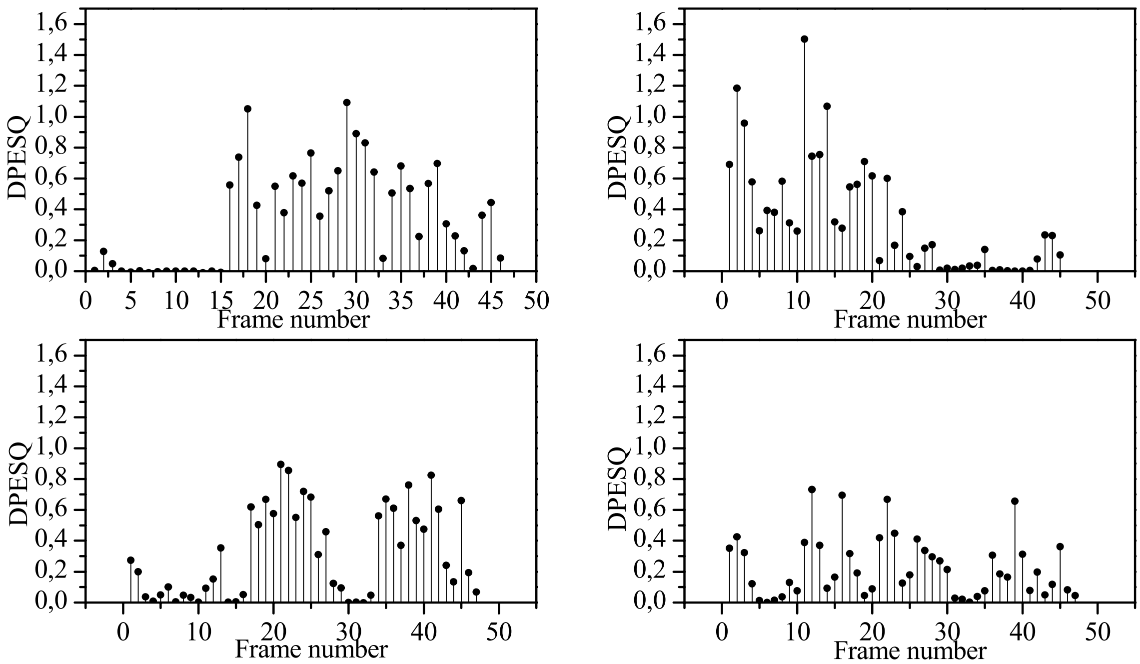


Figure 5: DPESQ as function of position of lost frame

From graphs it is obvious that speech quality after lost packet substitution substantially depends on loss position. There are places in a word where packet can be lost without noticeable speech quality degradation. Erasure of the packet in some other location decreases speech quality much more. We can see the same tendency in every investigated word. Different words are sensitive for losses in particular positions.

As it can be seen from Fig. 5 many DPESQ values are higher then 0.5. It means that only one lost packet in a word may decrease voice quality so that user satisfaction changes to lower category.

When instead of one lost packet we erasure two or more consecutive packets in a word, speech quality degrades even more. This is expected result of course. It is natural that when we have some consecutive packets erasures there is no point in a word where delta PESQ has small value.

Then during a mobile call, the iQoS expressed as $Q_k(t)$ would constantly change, reflecting the actual QoS received by that particular user. Evaluation of the received quality may be done in reasonably short time period, e.g. half- or one second. As humans may tell a word in a half- or one-second interval, impairment of quality over that period could be well noticeable.

Analysis of Figure 5 shows that is not deterministic relation between location of erasure and speech quality. A set of DPESQ values looks like an array of random numbers.

For the purpose to obtain some statistically meaningful data the following experiment was executed. At first 50 different words were chosen. In every separate word $k$ $(k = 1, 2, ..., 10)$ frames were erased randomly and independently. For every word 20 realizations of erasures were generated. After erasure DPESQ scores were calculated. Conditional empirical cumulative distribution functions were calculated:

$$F(x \mid k) = P(DPESQ \leq x), \qquad (5)$$

when in the word $k$ packets are lost. In the Fig. 11 there are presented functions $F(x \mid k)$, where $k = 1, 2, 3, 5, 10$. In the same figure unconditional cumulative distribution function is shown (labelled as unconditional) with premise that distribution of the number of frame errors is uniform.

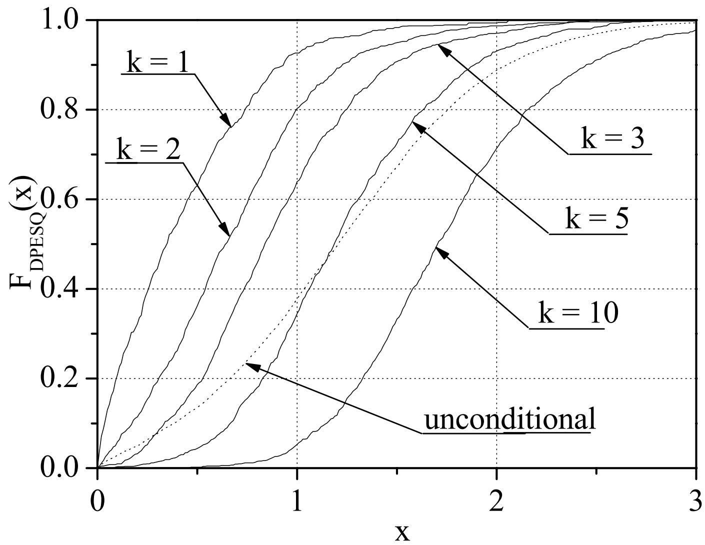


Figure 6: Empirical Conditional Cumulative Distribution Functions of Quality Degradation

Probability that DPESQ value will exceed $x$ is equal to 1-$F(x \mid k)$. If probabilities that $k = 1,2,3,...,K$ frames will be erased are known and equal $p(k)$ then cumulative distribution function may be expressed as:

$$P(\text{DPESQ} > x) = \sum_{k=1}^{K} p_k (1 - F(x \mid k)). \qquad (5)$$

From above formula it is clear that overall probability of quality degradation depends on the distribution of frame losses and on the conditional distributions of quality degradation values.

It can be seen that probability of DPESQ exceeding 1 is less than 0.1 when there is only one lost frame in a word. In such cases the meaning of word can be recognized easily. Situation becomes more complicated when number of errors increases. Sometimes word may become not intelligible and it is the most serious impairment.

## 5. Impact of lost frames on data transmission

Estimation of QoS dynamics of data transmissions is directly depends on capacity. However, the main protocol for data communications is Transmission Control Protocol. [14] TCP is a reliable, connection-oriented, full-duplex, transport protocol widely used in wired networks. Widespread mobile wireless environments are prone to packet losses, high bit error rates, and mobility induced disconnections.

TCP interprets packet losses as an indication of congestion and inappropriately invokes congestion control mechanisms, which leads to degraded performance [14].

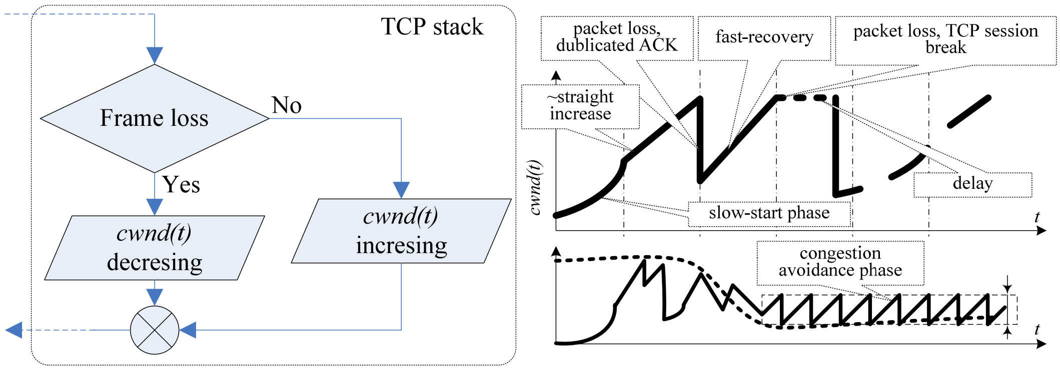


Figure 7: TCP Congestion window algorithm

Normally, in the wired networks the size of the congestion window is being periodically increased till it reaches its maximal window size ( $cwnd_{max}$ ) (Fig. 7), which is depends on capacity of the link [14]. In the most usable TCP Reno algorithm the link capacity depends on threshold time during which the confirmative of successfully frame transmission packet (ACK) must be received. The threshold time is named RTO:

$$RTO = \overline{RTT} + 4\sigma_{\overline{RTT}}. \quad (6)$$

There $\overline{RTT}$ – the average of the time needed to successfully transmit the packet; $\sigma_{\overline{RTT}}$ – the root-mean-square of deviation of $\overline{RTT}$ duration. If during this period ACK is not received, congested packet loss is detected. After that the decreasing of congestion window is unavoidable.

If information is transmitted via the widespread wireless link, where the packets are lost at random, the minimal congestion window size will be periodically adjusted [14]. Therefore, the capacity of TCP session will decrease. These suffer degradation of data transmission and capacity of the link.

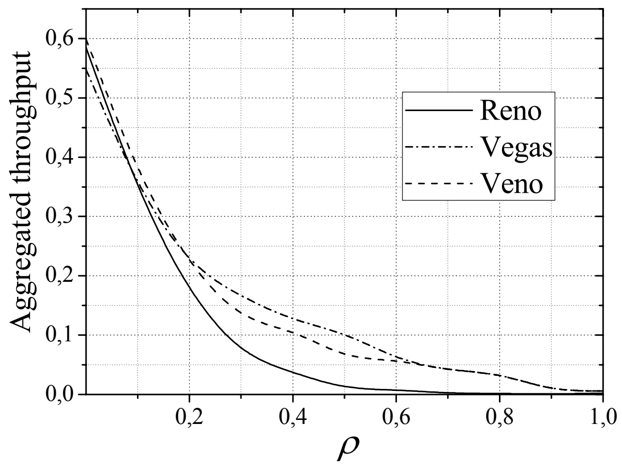


Figure 8: Aggregate Throughput vs. Random Packet Loss ($\rho$)
in the IEEE 802.11b network environment

Modelling results of TCP algorithms, published at [14] have showed (Fig. 8) that the degradation of TCP performance is not depending on choice of TCP algorithm implementation. It strongly depends on link packet loss probability. As it is showed in Figure 8 the degradation of aggregate throughput is distinguished if probability of packet loss is between 0.01-0.1. In such situation degradation of aggregate throughput is approximately 10%. If probability is more than 0.1 the throughput degradation is extremely high. However, we may come to a conclusion that in widespread wireless networks the packet losses influence the aggregate throughput of data transmissions. These suffer the degradation of capacity of wireless link, herewith making downgrading of the iQoS.

From that point of view the rating of tariffs for provided iQoS of data transmission must depend on successfully transmitted count of frames from ISP to end-user in the conventional period. It means that iQoS must directly depend on real data throughput in end-user side. We will investigate these questions more dentally in future.

## 6. Relations between SP and end-user

Relations between the Service Provider and the end-user as well as their interminability are regularized by means of contractual obligations. Currently wide spread rule is one that does not care about quality at all. Technical means of accounting calculates time (or amount of data) and knows nothing about quality. User also has not objective facts about perceived quality. Currently in practice user pays for service that assumed to be good and SP is not responsible if user is not satisfied with perceived service quality.

In next-generation networks SP and the end-user contractual obligations should include clause about quality level. For this purpose ITU formulated recommendation E.860. In this recommendation are stated general principles about Service Level Agreement (SLA). With the help of SLA it is possible to fully regulate relations between SP and the end-users. A SLA may include statements about performance, tariff rating and billing, service delivery. A SLA may include the compensations for an unachieved level of quality as an economic issue of the contract.

For SLA attitude implementation it is necessary to supply next-generation mobile telecommunication equipment with iQoS modules. iQoS modules should not only perform supplied quality evaluation, but also relate these evaluations with an amount of supplied service. For this reason quality modules should be integrated with pricing tools (Fig. 9). iQoS modules are necessary for SLA attitudes implementation between SP and the end-user. Gathered data could help users to protect their rights.

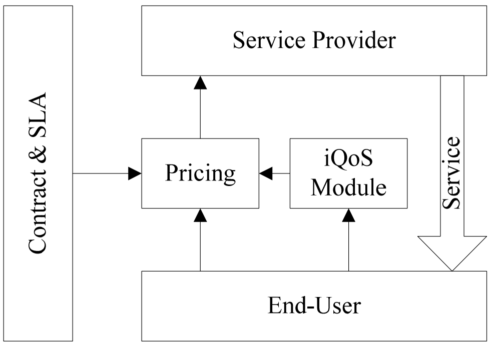


Figure 9: Support of SP and the end-user contract agreement

iQoS module collects all information required to evaluate the actual service received by a user, which can include poor service reception due to wireless channel impairment or unavailability as well as network congestion. This information then will be used to adjust any charge to a user.

In Figure 9 iQoS module is presented as a mean for controlling Service Level Agreement. For the purpose to implement SLA attitudes and relate them with tariffs it is necessary to solve many questions. The first group of questions concern how to describe relations between the end-user and the service provider.

When talking about tariffs, pricing – the first thing that is necessary to solve – to choose basic pricing unit - $u_0$. In conventional telephony basic pricing unit is duration of conversation expressed in seconds. Current and next-generation systems convey information in packets. Therefore it is reasonable to take as a basis the pricing unit packet of some form. This unit- $u_0$, obviously has and quantitative aspect – number of bytes $N_u$. Kind of service (voice, video, e-mail) can be marked with some kind of index. In this case index could be just number in list of services. More complex situation is with quality indicators.

Development of any current telecommunication technology considers quality requirements. When communication conditions are good and meets technological requirements, supplied quality level should be named nominal or required quality level - $Q_{os}^{req}$. This $Q_{os}^{req}$ quality level could be named reference quality level for iQoS evaluations. When SLA is constructed, required quality level $Q_{os}^{req}$ should be given as well as methods and algorithms for quality level calculation. For example, when voice is coded with AMR-n codec, quality of each codec differs [8]. Under real conditions perceived service level is equal or less than $Q_{os}^{req}$. The purpose of research is to find a set of quality degradation levels.

In order to specify quality levels, it is necessary to consider many factors: what quality impairments users will notice, if it is possible to evaluate these impairments precisely.

In order to solve some of the above mentioned problems revision of some well-established concepts should be performed. For example, according to ITU recommendations for subjective voice quality testing speech samples must be longer then 2 seconds. It is not obvious that such a long time interval is valid for iQoS measurements because average person can recognize intense impairment even when sample length is about 0.5s long.

Questions about compensation for unachieved quality level are even more complex. In this paper we are mentioned only part of questions related to SLA and iQoS modules.

## 7. Conclusion and future Work

1. Majority of quality evaluation methods have been created for the new codec's testing. All ITU recommended quality evaluation methods are based on assumption that communication conditions are stationary. This assumption is doubtless when impact of noise or disturbances is analyzed. Assumption about stationary conditions is extended to lost packets also. In this paper presented data shows that conditions of mobile communication are not the same for individual users. Sometimes these conditions become poor or even bad.

2. From the point of view of individual user conditions of mobile communication should be simulated as non-stationary. In some places of the network real service quality may become low compared to usual conditions.

3. ITU describes voice quality degradation levels in Opinion scale for Degradation Category Rating: inaudible, audible but not annoying, slightly annoying, annoying, very annoying. Non-stationary mobile communication conditions imply new category – bad voice quality – when parts of words or whole words are erased.

4. After proofing that conditions of mobile communication differs greatly we propose to make the next step – to create individual quality of service modules and their deployment system. Such module logically required for SLA system purpose to relations of ISP and end-user.

5. The criteria for RF power control are based on radio channel quality or received signal quality. It is important to notice that power control criteria are not identical to those used in voice or video quality determination.

6. This paper has investigated the effects of packet loss in single words. The results show that sometimes only one lost frame may impair speech quality hardly, still many lost frames sometimes has no impact on speech quality. Determined relation among the lost frames, series of packets, and degradation of directly comprehensible quality of voice isn't possible. This relation defined by irredeemable dependence of FUZZY.

7. Directly quality of voice with service tariff may be bind by traditional radio channel quality indexes: RxLev, CIR, and RxQual. In the GSM networks for example: if RxQual = 0, 1,…, 5 – we get highest QoS. Voice quality depends on real link conditions fractionally, because such mode is regulated by changing signal power. Threshold conditions RxQual = 7 is the particular zone of risk, in which offer many corrupted frames. The durations of cellular talk must by rated in reduced tariffs when the index of RxQual = 7 is determined (In observed networks such situation is 2-3 percents).

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