Delay Modeling for 3G Mobile Multimedia Services QoE Estimation

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Abstract. In this paper, we consider mobile multimedia services delivered over a UMTS network. In this convergent scenario, low level error recovery mechanisms of the access network entail that almost all packet losses are caused by frames arriving at the receiver later than the playout time. Our aim is to find a simplified yet realistic expression for the statistics of the end–to–end delay in order to characterize related application level losses and infer resulting Quality of Experience (QoE). We present a simple and easy to implement model based on empirically obtained parameters that quantitatively reflects the statistical properties of delay. We apply the model to estimate the behavior of application level losses and delay which are the key factors while forecasting QoE in VoIP. Obtained results prove the suitability of the model to be integrated in cross–layer adaptation mechanisms and its utility for dimensioning VoIP playout buffers is also tested.

Keywords: UTRAN, SR–ARQ, delay, QoE, VoIP.

1 Introduction

Nowadays the utilization of mobile communications for multimedia services over converged networks is becoming more frequent. For the reproduction of these streaming services, packets in the reception buffer are read at deterministically–spaced time intervals, so that those packets that do not respect the timing constraints imposed by applications (i.e. arrive later than expected playout time) are discarded. Therefore, at the playout time of these discarded packets either "nothing" is reproduced or just copies of previous samples, which negatively affects user Quality of Experience (QoE). So, these application level losses happen even in allegedly lossless networks, even under not–severe degradations, whenever the end–to–end (e2e) delay of a packet is above a playout time related threshold (implemented with a buffer). Non–interactive multimedia services usually allocate long buffers in order to cope with these losses, so that it is the jitter and not the delay the indicator to consider. Unfortunately, VoIP services can not use so long buffers due to associated initial delay (see [1]) since it would affect the interactivity of the conversation. Even with dynamic adaptative buffers the tradeoff between delay and losses imposes a maximum size for the playout buffer.

J. Del Ser et al. (Eds.): MOBILIGHT 2011, LNICST 81, pp. 67–80, 2012.

Then, our analysis aims at determining the probability that the e2e delay is above the threshold related to player buffer length, $Pr\{d_{e2e} > d_{max}\}$, which will provide us with application level loss rate. As a result, we need the Complementary Cumulative Distribution Function (CCDF) of the e2e delay to be evaluated at that threshold point instead. The scenario considered in this work includes a traditional 3G cellular access (UMTS) and a wired CN (Internet). However, the core part of the UMTS network (i.e. SGSN/GGSN nodes) and the Internet are assumed to introduce a constant delay, so the delay caused by low level error recovery mechanisms in the last mile radio part is the only contributor to the total variable e2e delay.

We have considered UMTS and not a HSPA/LTE/WiMAX scenario because it is the most widespread commercial network nowadays. Furthermore, unlike variants such as HSPA/HSPA+ where the final QoS (Quality of Service) will depend on the number of users and the scheduler, classical 3G connections provide dedicated channels (DCH), so that QoS constraints can be fulfilled. Additionally, the analysis of delay will be focused on ARQ (Automatic Repeat reQuest) based low level error recovery mechanisms that are quite similar to those in Beyond–3G (B3G) technologies.

Hence, the main objective of this work is to obtain an expression of the e2e delay CCDF under different reception and UMTS network conditions. To obtain the analytical expression of the delay statistics in the UTRAN (UMTS Radio Access Network), an in-depth analysis of retransmission techniques in UMTS has been carried out together with empirical fitting of simulation data. The impact of particular network and service parameters such as Block Error Rate (BLER), packet size, bitrate and average burst length (ABL) has been taken into account. Estimation of the delay calculated with the obtained model has been later used for predicting QoE on mobile VoIP service. Such closed–form solution will allow researchers to model network effects in cross–layer adaptation systems and design decision making algorithms accordingly.

The rest of the article is organized as follows. Section 2 describes retransmission schemes for UMTS Acknowledged Mode (AM). In Section 3 we propose using empirical data and fitting techniques in order to overcome the drawbacks of classical ARQ analysing schemes. Resulting expressions and fitted coefficients will be calculated just once. Later they will be included in a delay estimation system that will interpolate the obtained model to every possible input situation. Finally, Section 4 will show the capability of the proposed method for forecasting QoE for VoIP calls and Section 5 will summarize main contributions.

2 Preliminary Analysis of Delay for UMTS AM Mode

In this Section we will describe the analysis of the additional delay caused by retransmissions due to level 2 error recovery mechanisms in UMTS networks. Concerning the transmission in the UTRAN, different ARQ techniques are used in order to face the high error probability in the wireless medium, resulting in erroneous packets retransmissions. These mechanisms introduce a random delay at RLC (Radio Link Control) level, which is translated to a random delay at the application layer.

In UMTS networks, Selective Reject (SR) ARQ (see for example [2]) schemes are used instead of the typical ARQ ones (like Stop & Wait and Go-Back-N), due to their superior throughput performance. The basic operation of SR-ARQ consists on performing selective retransmission of frames, this is, only erroneous frames are retransmitted. These retransmissions happen when the sender receives a "NACK" (or Bitmap ACK with erroneous packets) belonging to an erroneous frame or when a timer reaches its time-out.

As defined in 3GPP Specifications [3], the RLC layer operation mode responsible for handling the frames recovery by means of SR-ARQ is called AM (Acknowledged Mode). As specified in this standard, error correction is done at MAC PDU (Medium Access Control Protocol Data Unit) or Transport Block (TB) level. In typical configurations these blocks have a payload capacity of 40 bytes, in which data from superior layers can be inserted, plus the corresponding bytes of RLC+MAC headers. In this way, upper level packets are segmentated in blocks of 40 bytes that are later on individually transmitted. TBs are transmitted at time intervals called TTI (Transmission Time Interval). One or several PDUs can be transmitted in each TTI, and it is the MAC layer which decides how many PDUs must be transmitted in each TTI (a typical example of bearer configuration is: 12 TBs transmitted per TTI and 10 ms TTI duration).

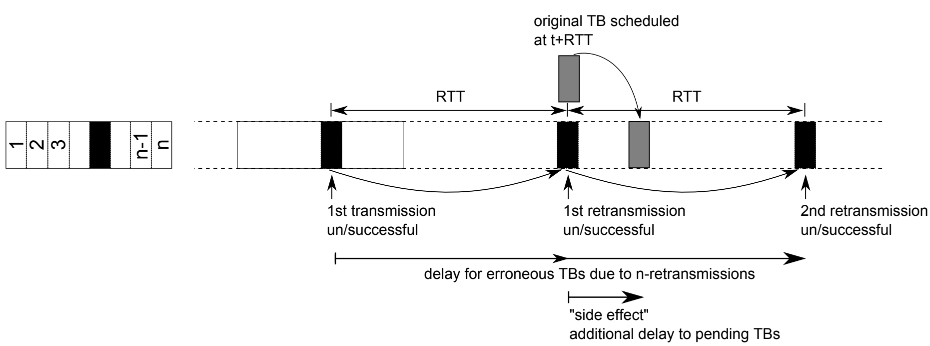

Fig. 1. ARQ retransmission from the sender point of view

Normally a RLC level transmitter uses both a transmission and a retransmission buffer. When a PDU is scheduled for submission, it is placed in the transmission buffer, and once transmitted to the channel, it is moved to the retransmission buffer. The PDUs in the retransmission buffer are deleted or retransmitted depending on the answer sent by the receiver. Retransmission buffer has priority over the transmission one; this is, upon the reception of erroneous frame notification, those to be retransmitted must be sent before the one originally scheduled in the transmission buffer, which has to wait to the next TTI (considering that there is no TB available in the RLC frame). Furthermore, since some feedback from the receiver is needed for the sender to detect the loss/error, possible retransmissions will arrive at the receiver delayed $n \cdot RTT_u$ ms., considering this UTRAN Round Trip Time (we will use the term RTT_u from now on) as the time needed for the "NACK" to go back to the sender plus the time needed for the retransmitted AMD (Acknowledged Model Data) PDU to arrive at the receiver (i.e. 100ms for typical UMTS commercial configurations). The overall processs is depicted in Fig. 1.

In the receiver, AMD PDUs remain in the reception buffer until a complete RLC SDU (Service Data Unit) is received. However, if a retransmitted RLC PDU is once again lost, retransmissions are tried until either the SDU Discard Timer expires or the number of retransmission exceeds a threshold (see different retransmission modes in [3] and [4]). Once all the PDUs associated to a RLC SDU are correctly received, they are reassembled and passed to superior layers. When all the PDUs of a RLC SDU are correctly received, resulting SDU is moved forward to the upper layer regardless the status of the previous RLC SDUs. This reception buffer policy is called not-in-order delivery. On the other hand, if SDU order is respected and SDUs with lower sequence number are required to be correctly received before sending data to higher layers, the buffer policy is in-order delivery.

Finally, due to the characteristics of the propagation of radio signals over the air, errors usually occur in bursts, and a correlation exists among them also at TB level in UTRAN. In [7] this behavior is modeled by means of modified Markov chains. For a dynamic scene (with mobility), it is concluded that the behavior of error bursts at TB level is similar to those of burst at TTI level, and this is modeled with a modified Gilbert–Elliot model (see [5] and [6]) at TTI level. On the contrary, for the static case a lower granularity analysis at TB level is required resulting in a more complex Markov model. In this context, burst length concept is defined as the number of consecutive TTIs or TBs with errors. These advanced models represent with higher fidelity the behavior of the wireless link, far beyond typical ones based on independent losses.

As a conclusion of this preliminary analysis note here that in our study we will consider UMTS RLC AM mode, SR-ARQ retransmission scheme, with 12 TB/TTI configuration, not–in–order delivery and complex bursty air error model in mobility based on modifications of Gilbert–Elliot's one in [7]. These considerations are not trivial since many studies do not state them explicitly and usually yield inaccurate simulation results. A common example is the UMTS module for ns-2 of the Eurane project [8] that does not support not–in–order delivery model, so that the evolution of the delay is not the expected one.

3 UMTS Delay Modeling

Once the behavior of the AN (Access Network) considered in our study has been analyzed we focus on modeling SR-ARQ.

In [9] a simple UTRAN SR-ARQ delay modeling is carried out based on combinatorics both for in–order and out–of–order delivery. A discrete PDF

(PMF-Probability Mass Function) evaluated only at RTT_u multiples is obtained. However, obtained expressions are too simplified solutions. On the one hand, they do not consider the typical situation where TBs are scheduled for transmission at the same time slot that a retransmission request. In this case, the slot would be occupied and therefore scheduled TB should wait to the next free TTI to be sent, resulting on an impact into delay statistics (see Fig. 1). Modeling this effect is not trivial. On the other hand, TBs are considered to be lost independently, which is also a simplification. In fact, as explained before, a correlation and a bursty behavior exists, which is not modeled by the authors.

Similarly, the analytical expressions proposed in [10] regarding SR-ARQ delay in the UTRAN, for the two reception buffer policies, are deduced on a greedy source situation, where packets are sent continuously, and later validated with simulation. However, the greedy source assumption is not always realistic. Furthermore, related mathematical expressions are based on iterative computation, so that obtained results are quite complex in order to be implemented in real mobile devices.

Considering the inability of different theoretical proposals for modeling delay in the UTRAN we decided to obtain a model of UTRAN SR-ARQ delay by analizing delay distributions out of simulation results.

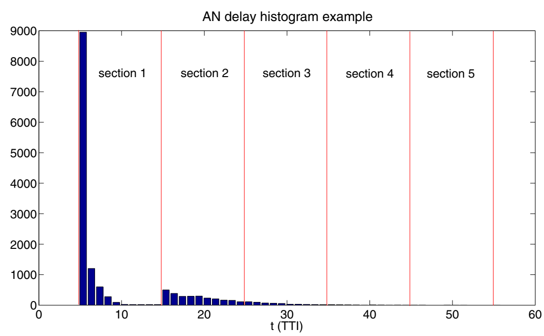

Fig. 2. Retransmission sections in an d_{AN} histogram

The basic analysis of the AM RLC mode and obtained simulation results for AN delay show a RTT_u-spaced periodic pattern due to successive retransmissions. Within each retransmission section most of the frames arrive at the first slot (TTI) available but there appear also delayed frames in successives slots due to the so-called "side effect" shown in Fig. 1. Then, it is clear that the solution must be defined in retransmission sections due to SR-ARQ working mode; where the first section represents the retransmission-free frames. When analysing simulation data, a RTT_u of 100ms is considered, so that the distance between retransmission periods will be 10 TTIs (a TTI duration of 10ms is used). At very high BLERs or bitrate situations, it is possible to find samples from previous section (so $i-1$ retransmissions) that appear in the following one (i^{th}

retransmission section) because they have been delayed more than a whole retransmission interval. This situation appears whenever the retransmission buffer is so full that no free slot is available for the originally scheduled ones for a whole RTT_u period. However, the contribution of those samples is considered negligible, so that this effect is not taken into account and the retransmission sections are considered as a single function for the whole retransmission period (note the regular function in (1)). Besides, there are no samples between the origin ($t = 0$) and the first sample in the first section. This delay is due to fixed transmission delay and packetization and reception buffer effects, and it is not taken into account for fitting. For convenience we shifted all delay samples to the origin, considering this the arrival of the first packet without losses. Fig. 2 shows the per-retransmission section behavior along up to 50 TTIs.

In order to obtain both the shape and empirical parameters for every network state and application setting considered an extensive set of simulations have been carried out. The main features about the simulation scenario are collected in Table 1.

Table 1. Simulation parameters for AN modeling

Element	Value
Loss model	Modified Gilbert Elliot's in [7]
Reception policy	Not-in-order delivery
Discard Timer (ms)	500
Bearer configuration	(384kbps DL, 64kbps UL)
TTI (ms)	10
Transport format (TB/TTI)	12
Transmission channel type	Dedicated Channel (DCH)
Simulation tool	OPNET Modeler

For PMF fitting, nonlinear Least Squares method [11] is used, which minimizes the summed square of residuals. We have modeled the PMF of the delay in the AN, $D_{AN}[n] = D_{AN}(t)|_{t=n\cdot TTI}$), by an approximation with delayed exponentials, so that the resulting analytic expression of the PMF of the delay in the UTRAN is shown in equation (1).

$$\sum_{i=1}^{N} a_i \cdot e^{b_i\cdot(t-(i-1)\cdot RTT_u)} \cdot \sqcap_{RTT_u}\left(t - i \cdot \frac{RTT_u}{2}\right) \tag{1}$$

where:

- $\sqcap_{RTT_u}\left(t - i \cdot \frac{RTT_u}{2}\right)$ is the rectangular function
  - width RTT_u
  - centered in $i \cdot \frac{RTT_u}{2}$
- i identifies the retransmission section.
- N is the maximum number of retransmission sections.
- a_i and b_i parameters are the result of the exponential fitting done with each section samples.

To validate the fitting (i.e. to check the correlation between the normalized histogram and the estimated delay PMF for each configuration) the R-Square parameter has been analyzed. Chosen exponential functions for all the retransmission sections provide good results (see Table 2).

Table 2. R-square values for packet size=30bytes, bitrate=12.2kbps, ABL=1.75 and different BLERs

BLER (%)	R-square (section 1)	R-square (section 2)	R-square (section 3)	R-square (section 4)	R-square (section 5
1	0.999	0.837	-	-	-
5	0.999	0.717	0.726	-	-
10	0.988	0.854	0.953	-	-
20	0.994	0.926	0.974	0.701	0.633
30	0.983	0.797	0.919	0.848	0.567

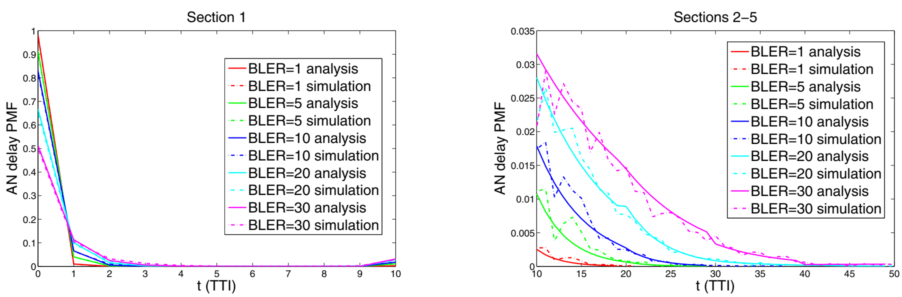

Fig. 3. Comparison between empirical and approximated AN delay PMF for packet size=30byte, bitrate=12.2kbps, ABL=1.75 and different BLERs

As shown in Fig. 3, the impact of BLER on the delay is clearly reflected, as when this is higher, retransmissions are more probable resulting in lower values for the histogram at the beginning of the first section (free of retransmissions). This behavior is also reflected on the values of a_i and b_i coefficients. Besides, the analysis of Table 3 shows that the fitting mechanism provides all the coefficients (including $i = 5$) at high BLERs only, since there are no delay samples for low BLERs in latter sections.

After carrying out the simulations with all possible combination of interesting input values (BLER, ABL, packet size and bitrate -see Fig. 4-) we have built a set of tables of a_i and b_i coefficients. Later, by using the expression (1) and a_i and b_i associated to the specific network state under study we can obtain the model for $D_{AN}[n]$ that will be finally used for the estimation of e2e delay. These lookup tables and the expression for $D_{AN}[n]$ provide researchers with a simple but effective method for forecasting the behavior of a UMTS network for certain RLC configuration sets and network states.

Table 3. a_i and b_i coefficients at the i–th retransmission section for different BLER (packet size=30bytes, ABL=1.75, bitrate=12.2kbps)

BLER(%)	a_1	a_2	a_3	a_4	a_5	b_1	b_2	b_3	b_4	b_5
1	0.98	2.5E-03	—	—	—	-458.97	-38.53	—	—	—
5	0.89	10.5E-03	6.0E-04	—	—	-312.54	-28.30	-35.88	—	—
10	0.79	0.017	2.6E-03	—	—	-253.49	-18.49	-33.12	—	—
20	0.62	0.026	8.2E-03	7.0E-04	2.0E-04	-189.40	-12.58	-23.75	-15.97	–20.36
30	0.46	0.029	0.013	3.0E-03	2.0E-04	-151.04	-7.61	-11.54	-14.84	–2.01

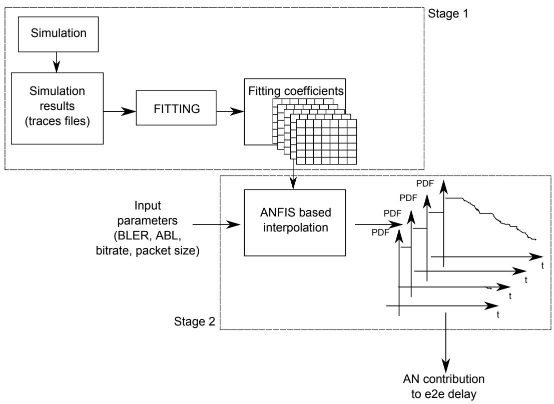

Fig. 4. Proposed process for modeling AN delay

Although a limited set of discrete input values for the AN modeling will be enough for most services, since they allow a limited range of values for bitrate or packet size, other multimedia services may require a full estimation of e2e delay for continuous ranges of service and network states. For these special cases, we propose a second stage consisting of an ANFIS (Adaptive Neuro-Fuzzy Inference System) [12] [13]) based interpolation mechanism to obtain the estimation for all posible combinations of inputs. Note here that, regardless the granularity needed, the first stage has to be carried out just once. After the ANFIS system is deployed and trained, it will provide the estimation of the delay distribution for any possible input parameters (BLER, ABL, bitrate and packet size) with no need of additional modeling. We have developed an ANFIS–based learning model to predict a_i and b_i values for different combinations of BLER, ABL, packet size and bitrate. In our system, after training the ANFIS network with simulation–derived dataset, we have obtained a nearly "infinite" interpolated shape for different a_i and b_i parameters of the exponentials in (1).

In Fig. 5 we can see the comparison between the original, empirical data-based approach and the ANFIS extended one for different combinations of BLER and packet sizes.

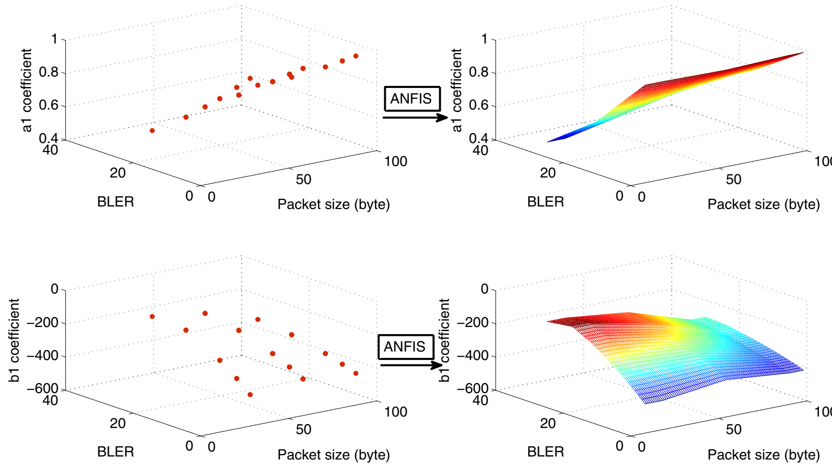

Fig. 5. Extension of a_1 and b_1 using ANFIS

As a conclusion, in comparison with previous studies about the SR-ARQ in UMTS networks, we have obtained a much simpler expression of the PMF but still capable of providing a good approximation of UTRAN behavior for different BLERs (and different packet size, bitrates and lower level parameters). Nevertheless, in order to know when the delay exceeds a threshold the CCDF is needed. Formally,

$$CCDF\left\{d_{AN}\right\}(t) = 1 - \sum_{k=0}^{\left\lfloor \frac{t}{TTI} \right\rfloor} D_{AN}\left[k\right] \tag{2}$$

As the $AN\ PMF$ is defined on the interval $\left[0, N \cdot \frac{RTT_u}{TTI} - 1\right]$ the value of k is limited, so that k_{max} is:

$$k_{max} = N \cdot \frac{RTT_u}{TTI} - 1 \tag{3}$$

As was expected, since PMF approximation was good, obtained CCDFs match those obtained with simulation (see Fig. 6).

Moreover, recall that the fixed delay (d_f) introduced by propagation, serialization and forwarding processes both in the wireless and wired segments is not included in the previous expression. As obtained from simulation, this delay is about 60ms. Therefore, the e2e delay CCDF is computed as follows:

$$CCDF\left\{d_{e2e}\right\}(t) = \begin{cases} 1 & t < d_f \\ CCDF\left\{d_{AN}\right\}(t - d_f) & t >= d_f \end{cases} \tag{4}$$

This expression will be used in next section to infer QoE in VoIP.

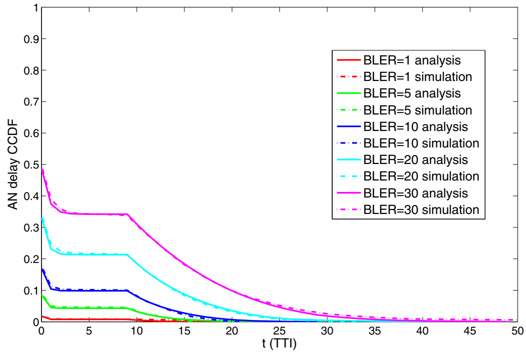

Fig. 6. Comparison between modeled and simulated AN delay CCDF for packet size=30bytes, bitrate=12.2kbps, ABL=1.75 and different BLERs

4 Forecasting VoIP QoE

Finally, we will show the capability of the proposed system for forecasting the evolution of the VoIP QoE under different network states. AMR-based VoIP services [14] will be analyzed, comparing the obtained analytical results with the simulated ones.

The predominant method for assessing the QoE of VoIP services is the E-model, defined by the ITU-T in [15] and applied to AMR-based VoIP services over 3G UMTS connections in [16] resulting in expression (5).

$$R = R_o - I_s - I_d - I_{e-eff} + A \tag{5}$$

This model allows computing the value of a rating factor, R, which provides an evaluation of the communication impairment, and relating this score to the corresponding MOS (Mean Opinion Score) score. The additional impairment due to the network performance is caused by two main factors: I_d is defined as the impairment due to the total mouth–to–ear delay, while I_e is the impairment factor due to the combined effect of the codec and packet losses.

For a specific e2e delay distribution, the values experienced by I_d and I_e factors will depend on the application buffer size. At high buffer sizes, user experiences a higher delay in the communication, and thus I_d increases. Yet, all those voice samples that reach the destination endpoint with a variable delay inferior to the buffer value are successfully presented to the end user, resulting on lower I_e impairments. On the contrary, lower buffer sizes will result on lower I_d values but likely increased I_e impairment, depending on the e2e delay statistics. Therefore, the proposed e2e delay estimation method is worth to estimate the expected QoE at different network states, as well as to infer the most suitable buffer policy for each case.

Table 4 represents the estimated MOS scores for a concrete VoIP configuration (AMR–12.2 and 1 voice frame per IP packet), using a dejittering buffer of

200ms, for different combinations of network states (represented by the experienced BLER value at the radio segment). In order to validate the proposed QoE estimating approach, two MOS values (MOS_{VoIP}) are provided for each network state. The first one illustrates the outcome of the proposed model: the e2e delay CCDF is computed with (4) and the specific application–level delay and loss ratio is obtained for the established buffer size. The second value is the average MOS as directly computed from simulation traces. As can be observed, the proposed system provides good estimations of the expected values of the QoE for this kind of VoIP services.

Table 4. Forecasted and simulated MOS values for VoIP AMR calls (MOS_{VoIP})

BLER (%)	MOS_{VoIP} Model	MOS_{VoIP} Simul.
1	3.84	3.78
5	3.50	3.70
10	2.91	3.01
20	2.07	2.14
30	1.54	1.55

The expressions presented in this paper can be further used for optimizing the provisioning of multimedia content in convergent networks. Once identified the current network state for a VoIP user, it is easy to check the enhancement in terms of QoE of possible cross–layer adaptations that will lead to a different state.

Additionally, the method proves to be useful for real–time buffer dimensioning also. As illustrated in Fig. 7, once obtained the e2e delay CCDF for the specific case, estimating the associated application–level losses for different buffer size values is straightforward. Later, obtained new application–level delay and loss ratio values can be taken as inputs for the MOS estimation expressions, in order to infer the most suitable configuration for the current state. According to Fig. 7 we can conclude that, for that particular network state, a buffer of 100ms is better in terms of QoE than a buffer of 150ms since the CCDF is almost identical (so that application level losses -and I_e- are the same) while the constant delay due to buffering (included in I_d) is higher on the second case.

Therefore, reception buffer dimensioning plays an important role in order to obtain the best service configuration for a certain network state, and our analytical results provide finding the most appropriate buffer length for each case. The expected MOS map for different network and service states is summarize in Fig. 8. Network states are defined with BLER values 1, 5, 10, 20 and 30, while service state, s, is defined as:

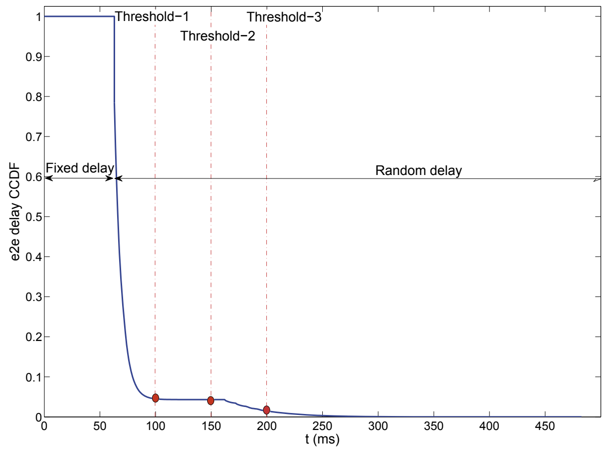

Fig. 7. CCDF of delay and associated impact into MOS_{VoIP} of different delay thresholds

$$s = (i - 1) \cdot 123 + (j - 1) \cdot 41 + k \tag{6}$$

where:

- AMR mode: i = (1 - 3) for 12.20, 10.20, 7.95 kbps.
- Packetization: j = (1 - 3) for 1, 2, 3 frames/packets.
- Buffer size: k = (1 - 41) for 100, 110, 120, ..., 500 ms of buffer.

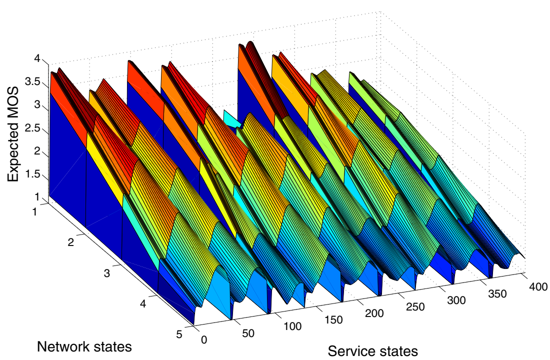

Fig. 8. Expected MOS for different network and service states

The VoIP configuration will be represented as a 3-tuple of parameters (i, j, k). For all the network states defined, we can find the most suitable VoIP configuration for each considered network case. Table 5 summarizes the results obtained in the whole process. As was expected, the best service configuration for all network states is the AMR–12.20 codec mode with one voice frame per packet. However, if the distance in the MOS scale between two possible VoIP configurations is low, it could be preferable the configuration with a lower bitrate requirement; this is, with lower AMR mode or higher packetization scheme. This is useful from the standpoint of the optimization of resource and battery usage. On the other hand, different optimum dejittering buffer sizes are obtained for each network state, which clearly reflects the importance of using an adaptative buffer.

Table 5. Best choice of service level parameters

Network state (BLER %)	Optimum service state	Expected MOS
1	(1,1,180)	3.90
5	(1,1,210)	3.51
10	(1,1,270)	3.21
20	(1,1,320)	2.72
30	(1,1,370)	2.23

5 Conclusions

In this paper, we propose a model for estimating e2e delay aimed at estimating QoE in multimedia services over 3G networks. This way, application level losses due to excessive delay have been considered.

The main contribution of this paper is that a simple yet realistic model for e2e delay statistics has been described. A new retransmission–based empirical model has been proposed for the analysis of the delay statistics in the UTRAN. This model overcomes the limitations found in the state of the art, and allows estimating the random delay contributions at different service and network conditions. We have obtained both a fitted closed–form expression, for coarse grained network-service states, and ANFIS based estimation for finer granularity.

Finally, e2e delay characteristics have been mapped to user experienced QoE for the specific case of AMR VoIP service using the E–model. Estimated MOS_{VoIP} values fit simulation results and make our system suitable to be included in decision makers of multimedia cross–layer adaptation systems. This way, by forecasting MOS_{VoIP} for the network state that would result after every possible adaptation, we can choose the best suitable combination of service level paramaters and network states. For the same network state, the model allows us to calculate the length of the buffer associated the most convenient tradeoff solution between losses and delay.

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