DESIGN AND EVALUATION OF A LINK ADAPTATION ALGORITHM FOR ENHANCED VIDEO SERVICE PROVISION WITHIN MOBILE RADIO ENVIRONMENTS

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Abstract - This paper presents and evaluates a Link Adaptation algorithm for real-time video transmission within a mobile radio environment. The aim of the algorithm is to improve performance not only in terms of throughput but also in terms of delay. The results show that the objectives set by the proposed algorithm are more effectively fulfilled compared to more traditional algorithms aimed at system throughput optimisation.

Keywords – Link Adaptation, radio resource management.

I. INTRODUCTION

The evolution of mobile communications is being accompanied by the introduction of new multimedia services, such as video streaming and video conferencing. The bandwidth requirements of such services create new challenges to operators that need to introduce not only new radio interfaces but also to implement the means to efficiently manage the scarce available radio resources. The efficient use of radio resources is achieved by means of Radio Resource Management techniques such as Link Adaptation (LA). The potential and benefits of LA are such that it is now considered as a key technology for evolved 2G systems.

The basis of LA is to assess the channel conditions and then use a transport mode, from a set of predefined options, that is optimised for these conditions according to a predefined criteria. A criteria commonly used in the literature is to select the transport mode that maximizes the system throughput [1]. Although this criteria might be appropriate for best-effort services, its suitability could be questioned for real-time services with tight transmission delay and error performance constraints. A different approach is proposed in [2], aiming at achieving a target error probability, in particular for a music streaming service. The LA algorithm proposed in [3] is designed so that the transport mode selected maximises the video quality measured in terms of the Peak Signal to Noise Ratio. As the long-term throughput does not quantify the quality of service experienced by the user for delay-sensitive services, the work reported in [4] proposes a LA algorithm that aims at reducing transmission delays. Since the packet transmission delay depends on the size of the packet to be transmitted, a transport mode selection algorithm that is dependent on the packet size is presented in [4]. The aim of this paper is to introduce and evaluate in a dynamic environment, a new LA algorithm that also seeks to reduce the transmission delay but by directly including such performance measure in the adaptation process.

II. SIMULATION ENVIRONMENT

A. General Packet Radio Services (GPRS)

The work presented in this paper is based on the GPRS radio interface, which offers four different coding schemes - see Table 1 below. These Coding Schemes (CS) offer a trade-off between throughput and coding protection, paving the way for the application of dynamic LA to GPRS. Since GPRS uses a single modulation scheme, the LA algorithms considered in this paper will only adapt the CS employed.

Table 1. GPRS channel coding parameters

Scheme	Code rate	Payload	Data rate (kbits/s)
CS1	1/2	181	9.05
CS2	≈2/3	268	13.4
CS3	≈3/4	312	15.6
CS4	1	428	21.4

Although the low transmission rates achieved by GPRS may raise some questions about the feasibility of video traffic transmission within these systems, it is worth mentioning that the aim of this work is not to prove the feasibility of video transmissions over GPRS but to propose LA algorithms that improve transmission delays. To this effect, the availability of advanced link-to-system level interfaces is essential [5]. Such interfaces, described in Section II.C, are readily available for the GPRS system and support our choice to select this system to carry out the performance evaluation of the LA techniques proposed. Furthermore, this paper is confined to a single slot transmission scenario, since link level models available in the literature do not contemplate a multislot scenario. As reported in [6], link level models for multislot transmissions would be desirable, especially if the study of adaptive radio link techniques is considered. Once more, this scenario suffices to evaluate performance improvements between traditional LA algorithms and the proposed technique.

B. Simulation tool and system modelling

In order to ensure high accuracy and to account for sudden channel quality variations, an event-driven simulator working at the burst level has been implemented. The simulator models a sectorised macrocellular network and concentrates on the downlink performance. Users are assigned channels in a first-come-first-served basis and the channel is kept until all its data has been correctly transmitted. A single slot allocation strategy has been implemented by means of a random allocation scheme. Although mobility has been implemented, handover between sectors has not been considered. The main simulation parameters employed are summarised in Table 2. A full description of the simulation tool can be found in [7].

Table 2. Simulation parameters

Parameter	*Value*
Cluster size	4
Cell radius	1km
Sectorisation	120º
Modelled interference	1st and 2nd co-channel tiers
Nº of modelled cells (wrap-around)	25
Slots per sector	16
Users per sector	12
Traffic type	H.263 video: 6 users/sector WWW: 3 users/sector Email: 3 users/sector
Pathloss model	Okumura-Hata
Shadowing	Log-normal distribution 6dB standard deviation and a 20m decorrelation distance
Vehicular speed	50km/h
ARQ protocol	Only for WWW and email users. Assumed: perfect feedback of ARQ report and no RLC block losses
ARQ window size	64 RLC blocks
ARQ report polling period	16 RLC blocks

C. Link-to-System level interfaces

In order to reduce the complexity of system level simulations, the effects at the physical layer are generally included by means of Look-Up Tables (LUTs). Following the indications provided in [5], a set of advanced link-to-system level interfaces working at the burst level have been considered. As illustrated in Figure 1, this interface is composed of two LUTs. The interface requires as input from the system level the mean CIR experienced in a given burst. LUT-1 extracts the burst link quality, represented by means of the Bit Error Rate (BER), for the measured burst CIR. LUT-1 represents a Cumulative Distribution Function (CDF) of the BER for a given CIR. A random process is then used to generate the actual BER from the corresponding CDF. The interest of this procedure is to model the effect of fast fading on the BER through a random process, thereby including the fast fading at the system level. The BER is then estimated for the four bursts used to transmit a RLC block and LUT-2 maps the mean BER and the standard deviation of the BER over the four bursts to a corresponding Block Error Rate (BLER) value. Examples of the two LUTs can be found in [7].

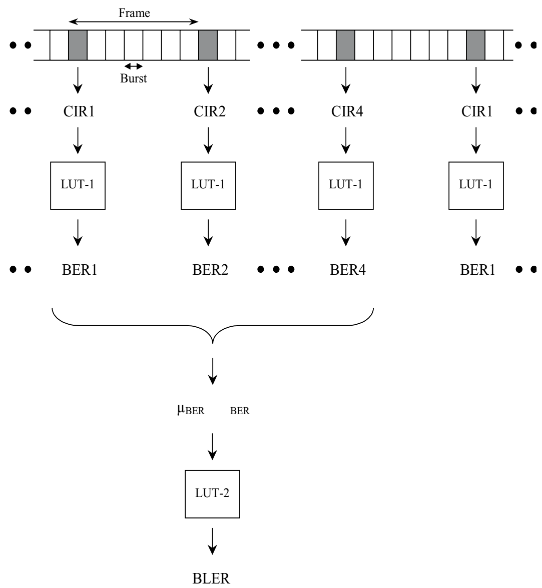

Figure 1 - Link-to-System level interfaces.

D. Traffic modelling

This work has considered three different traffic sources: H.263 video, email and WWW browsing. No channel partition has been applied between the different services. The WWW and email traffic sources have been implemented as an ON/OFF model [7]. For both traffic models, the transmission of a new packet cannot start until the previous transmission has finished, i.e. all the data has been correctly received. The active transmission time will hence depend on the link quality conditions.

The H.263 video traffic model considered employs three different frame types, namely I, P and PB, and targets a bit rate of 16 Kbit/s [8]. Each frame type exhibits different statistical properties, which are accurately captured by the model [8]. As depicted in Figure 2, the video traffic model considers the following variables and properties:

- *Frame size,* S_X. The correlation and distribution properties of each frame type need to be considered.
- *Frame duration,* T_X. I-frames are coded at deterministic rates. PB-frames code two consecutive frames as a single entity. Hence, frame duration is a relevant feature to be considered.
- *Correlation between frame size and duration.*

- *Frame transition rate.* P- and PB-frames are generated in bursts by the codec in order to sustain a particular target bit rate.

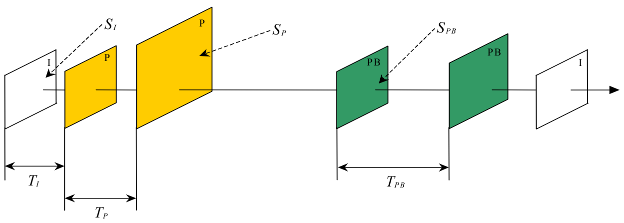

Figure 2 – H.263 traffic stream

The modelling is performed at two levels. On one hand, as illustrated by Figure 3, a Marchov chain drives the transition process between different frame types. On the other hand, the size, S_X, and duration, T_X, of each frame type, *{I,P,PB},* is determined by means of the following transformations

$$S_x = F_S^{-1}(F_x(X_0)) \qquad x \in \{I, P, PB\} \tag{1}$$

$$T_x = F_T^{-1}(F_S(S_x)) \qquad x \in \{I, P, PB\} \tag{2}$$

where X_0 is generated from a normal distribution with the estimated average and variance for the frame type considered. $F_X()$ is the X_0 CDF. $F^{-1}{}_S()$ is the inverse of the CDF estimated from the particular frame sizes available in the empirical trace.

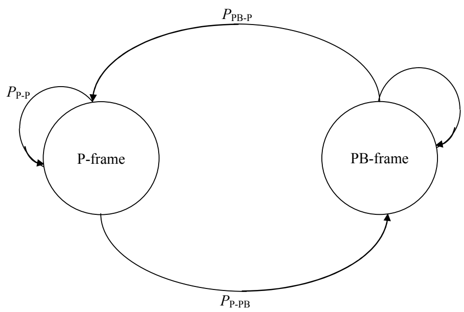

Figure 3 – H.263 frame transition Markov chain.

III. LINK ADAPTATION ALGORITHMS

The basis of LA is to adaptively select the optimum CS according to the channel quality conditions and a predefined criteria. This section presents two LA algorithms that differ on the criteria used to select the optimum CS.

A. Throughput-based LA algorithm

The LA algorithm proposed in [1], and commonly employed in studies considering the use of LA, tries to maximize the system throughput. As a result, the algorithm considers a CS to be optimum if it maximises the throughput defined as:

$$Throughput = R_{CS\text{-}i} \times (1 - BLER_{CS\text{-}i}) \tag{3}$$

where $R_{CS\text{-}i}$ and $BLER_{CS\text{-}i}$ are the data rate and BLER for a given CS respectively. Taking into account the particular link-to-system level interface considered in this work, an example of the throughput performance used to define the LA switching thresholds is illustrated in Figure 4. The LA switching thresholds define the boundaries between the regions where each CS is regarded as optimum.

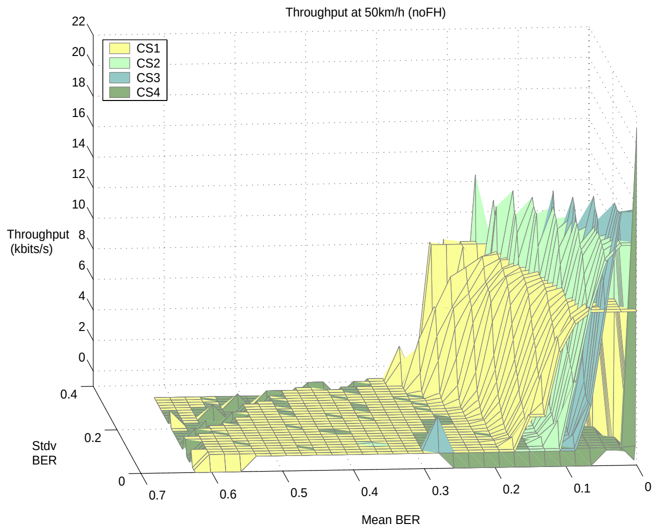

Figure 4 - Throughput and LA switching thresholds for the throughput-based LA algorithm

B. Proposed LA algorithm

Since real-time video transmissions have tight constraints in terms of delay performance, the LA algorithm proposed in this paper not only considers the throughput performance as the selection criteria but also the delay, defined as follows [9]. We assume that a video frame with size S is generated at time t_0 and that the next video frame will be generated at time t_0+D. The time needed to finish the transmission of the first video frame can be defined as t_0+D+δ, with δ being the *video frame transmission delay* defined in this paper. If δ>0, the transmission of the first video frame has not finished by the time the next video frame is generated. On the other hand, if δ<0, the transmission of the first video frame has finished before the next video frame is generated and the channel can be released so that other users can access it. In order to support real-time video communications, the transmission of the first video frame needs to be finished before the next video frame is generated. Since this work considers real-time video transmissions, the use of an ARQ protocol for such services is avoided. As a result, the video frame transmission delay depends on the data rate of each CS rather than on the system throughput. Hence, if a fixed CS is used for the video frame transmission, δ can be expressed as:

$$\delta = \frac{S}{R_{CS-i}} - D \tag{4}$$

where $R_{CS\text{-}i}$ is the data rate of the considered CS. From (4), improving the delay will require the use of CSs with a high data rate and therefore with less error protection.

In terms of the LA algorithm design, if δ>0, the algorithm should try to minimise δ, whereas if δ<0, the algorithm should try to maximise |δ|. As a result, and taking into account that the delay is not dependent on the throughput but on the data rate and if the aim is to maximise the system throughput while minimising the video frame transmission delay, the proposed LA algorithm should try, for each CS, to maximise (5a) if δ>0 or (5b) if δ<0:

$$\frac{Throughput}{Delay} = \frac{Throughput.R_{CS-i}}{S - D.R_{CS-i}} \quad (5a)$$

$$Throughput.\left|delay\right| = Throughput.\left(D - \frac{S}{R_{CS-i}} \right) \quad (5b)$$

with the throughput defined in (3).

Therefore, the operation of the proposed LA algorithm is as follows. For each video frame and every GPRS CS, the algorithm computes the delay as defined in (4). Depending on whether the delay is positive or negative, equations (5a) or (5b) are evaluated for each CS. Similar to the operation of the throughput-based LA algorithm, the result of each one of the previous evaluations for each CS is compared in order to decide the LA switching thresholds. It is worth noting that as the throughput is dependent on the BLER, both equations (5a) and (5b), and consequently the LA switching thresholds calculated, will also depend on the BLER and ultimately on the burst link quality conditions. An interesting and desired behaviour that has been observed in different simulations is that as the size of the video frames increases the use of the CSs with higher data rates also increases. This trend allows the transmission delay reduction at the expense of an increased risk of transmission errors.

Direct comparison of the throughput-based LA algorithm operation with the proposed algorithm reveals the necessary calculation of the LA switching thresholds for each video frame to be transmitted as a possible drawback. However, this drawback is greatly minimised for downlink transmissions since all the calculations and decisions will be made at the base station. Moreover, as it will be shown in Section IV, this potential drawback is overcome by the performance improvements that can be obtained with the proposed LA algorithm.

IV. PERFORMANCE EVALUATION

A. Performance measures

The performance of the two considered LA algorithms for H.263 video transmissions will be primarily assessed in terms of throughput and delay. To this effect, it is necessary to consider not only the mean performance but also other performance metrics, such as the minimum throughput or maximum delay experienced by 95% or even 99% of the samples, since they provide a better indication of the QoS supported by each algorithm. The throughput is defined as the total number of bits successfully transmitted over the air interface divided by the radio transmission time. The delay performance is evaluated by means of the normalized delay, which corresponds to the time needed to transmit a block of data divided by the size of such block. Other parameters of interest to understand the operation of the LA algorithms are: the usage percentage of each CS, the proportion of RLC blocks received with the optimal CS and the number of CS changes requested by LA. For both algorithms, the LA updating period, defining how regularly a decision is made on the most suitable CS, has been set to 60ms [7].

The quality of the received video frames is evaluated by means of the BLER and BER. In terms of the BLER, three different error rate targets are considered. A BLER below 5% would not produce a noticeable video degradation for H.263 transmissions [10]. On the other hand, a target BLERs of 1% and 2% are suggested in [11] for videophony and streaming/real-time video applications respectively. As a reference, the indications in terms of BER performance provided in [12] will be considered in this paper. In particular, a BER of 10^{-6} will provide no visible degradation, a BER of 10^{-5} will provide little visible degradation, a BER of 10^{-4} provides some visible artefacts and a BER higher than 10^{-3} provides little practical application.

Table 3. Performance comparison of LA algorithms

Parameter	*Algorithm*	*Mean*	*95%*	*99%*
Throughput (kbps)	Throughput	16.56	10.46	7.53
	Proposed	16.94	11.33	8.09
BLER (%)	Throughput	11.18	25.90	39.73
	Proposed	12.20	28.91	42.71
Normalized delay (ms/kbit)	Throughput	58.32	99.08	129.44
	Proposed	55.36	85.32	119.59

B. Numerical results

As it can be seen from Table 3, the proposed LA algorithm improves the throughput performance compared to the throughput-based LA algorithm. The improvement obtained is particularly significant for the most restrictive QoS parameters. It can be observed that the minimum throughput obtained for 95% and 99% of the samples is improved by 8.3% and 7.4% respectively, whereas the mean throughput improvement is 2.3%. These significant improvements are due to the increase in the percentage of RLC blocks received with the optimal CS - see Table 4. The better operation of the proposed LA algorithm also becomes apparent as the mean number of CS changes is analysed. For the scenario considered, the throughput-based LA algorithm requests 4.69 CS changes per second whereas the proposed LA algorithm only requests 3.94 CS changes per second. This figure not only demonstrates that the mode selection of the proposed algorithm was more accurate but also that the signalling load associated with the use of LA is considerably reduced (16%) with the proposed LA algorithm.

From Table 4, it can be observed that the proposed algorithm also increases the use of the less robust CS; this increase was foreseen in Section III-B. The drawback of using such CSs is the increase in the BLER observed with the LA algorithm proposed in this paper - see Table 3.

Table 4. CS percentage usage and percentage of RLC blocks received with the optimal CS

Algorithm	*CS1*	*CS2*	*CS3*	*CS4*	*Opt. CS*
Throughput	6.63	7.14	26.41	59.82	65.99
Proposed	3.37	2.92	27.98	65.73	71.17

Table 5. % of video frames transmitted with/without delay

Algorithm	*Without delay*	*With delay*
Throughput	69.16	30.84
Proposed	73.92	26.08

Table 6. Percentage of video frames transmitted without delay and that guarantee a given error rate target

	Throughput	*Proposed*
BLER ≤ 1%	44.73	46.22
BLER ≤ 2%	44.75	46.24
BLER ≤ 5%	44.98	46.45
BER ≤ 10^{-3}	45.93	48.10
BER ≤ 10^{-4}	40.40	42.14
BER ≤ 10^{-5}	39.33	41.02
BER ≤ 10^{-6}	39.23	40.92

In terms of the delay performance, the results illustrated in Table 3 also demonstrate the improvement obtained with the proposed LA algorithm. In particular, the mean normalized delay and the maximum normalized delay experienced by 95% of the samples are reduced in the order of 5% and 14% respectively. These improvements are a result of a more aggressive approach in terms of the mode selection since the use of the CS with higher data rates is considerably increased with the proposed LA algorithm. As shown in Table 5, the lower normalized delay obtained with the proposed algorithm results in an important increase in the number of video frames that are transmitted without delay, i.e. that their transmission is finished before the next video frame has to be transmitted in a real-time scenario.

Although the throughput and delay performance improvements achieved with the proposed algorithm were obtained at the expense of a higher BLER, the results shown in Table 6 demonstrate that a higher percentage of video frames transmitted without delay and with the required error rate targets suggested by [10]-[12] are still attainable with the proposed LA algorithm. The results obtained show that the proposed LA algorithm is a suitable candidate for improving the real-time transmission of video in mobile networks.

V. CONCLUSIONS

This paper has proposed a new LA algorithm designed to improve the performance of services with tight delay constraints. The algorithm bases the transport mode selection not only on the system throughput but also the transmission delay. The algorithm has been evaluated under a real-time H.263 video transmission and its performance has been compared to the commonly used throughput-based LA algorithm. The results obtained show that the proposed algorithm not only improves the delay performance compared to the throughput-based algorithm but also the throughput itself.

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