Link adaptation algorithm for improved wireless transmission of delay-sensitive packet data services

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Link adaptation is a radio resource management technique that selects a transport mode based on the experienced channel conditions. The optimum mode is commonly determined so as to maximise the throughput. Although, this approach is suitable for best-effort services, it is not tailored for real-time services. Presented is a new link adaptation algorithm designed to improve the transmission of delay-sensitive services. The results demonstrate that significant improvements in terms of transmission delay, throughput and operation of link adaptation itself, can be obtained with the proposed scheme.

The basis of link adaptation (LA) is to dynamically select, according to a predefined criteria, the optimum transport mode (e.g. modulation and/or coding scheme) based on the experienced channel conditions. One of the most accepted criteria is to base the selection on throughput [1]. Although this approach may be appropriate for best-effort services, real-time services have QoS requirements not only in terms of throughput but also in terms of transmission delay and error performance. As a result, a number of alternative algorithms have been proposed. A mode selection scheme based on the error performance was proposed in [1], while [2] presents an LA scheme designed to reduce transmission delays in a system using selective ARQ for error recovery.

Alternatively, this study concentrates on real-time services that might be forced to avoid using ARQ protocols because of the additional delays incurred by the retransmission of erroneously received data blocks. In this context, this Letter presents a new LA algorithm aimed at reducing the transmission delay and simultaneously maximising the throughput.

Reference link adaptation algorithms: This work is based on the General Packet Radio Services (GPRS) radio interface. GPRS considers a single modulation scheme but defines four coding schemes, CS1 to CS4. These coding schemes (CS) offer data rates ranging from 9.05 to 21.4 kbit/s, with CS1 being the most robust CS and CS4 the least robust. As a result, these CS offer a trade-off between throughput and error protection, paving the way for the application of LA to GPRS.

The commonly used throughput-based algorithm (TA) [1] regards a CS as optimum if it maximises the throughput defined by:

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

where $R_{CS\text{-}i}$ and $BLER_{CS\text{-}i}$ are the data rate and block error rate (BLER), respectively, for a given CS.

The error-based algorithm (EA) proposed in [1] is designed to reduce transmission errors and to achieve a particular target error probability. Since EA performance could significantly differ from the error target, [3] proposed a modified version of the EA algorithm (MEA) that obtained a performance close to the target error rate. The principle of MEA is to constantly evaluate the experienced average BLER, and based on whether such average BLER is above or below the considered target error rate, to allow or forbid the use of the less robust coding scheme. In this work, a target BLER of 5% and an averaging period of 1 s are considered for MEA.

The TA, EA and MEA algorithms will be used to benchmark the performance of the proposed scheme.

Delay-sensitive link adaptation algorithm: Let us assume that a video-frame with size S is generated at time t_0 and that the next video-frame is generated at time $t_{next} = t_0 + D$. To support real-time video communications, the transmission of one video-frame needs to be finished before the next one is generated. Let us define the time when the transmission of the first frame is completed t_{finish} as $t_{next} + \delta$. If $\delta > 0$, the transmission of the first frame has not finished by the time the next frame is generated. On the other hand, if $\delta < 0$, the transmission of the first video-frame has finished before the next one is generated at time t_{next}. Since we concentrate on real-time transmissions avoiding the use of ARQ protocols, the transmission time of a video-frame depends on $R_{CS\text{-}i}$ rather than on the throughput. Hence, if a fixed CS is used for the transmission of a frame:

$$\delta = \frac{S}{R_{CS-i}} - D \tag{2}$$

The delay-sensitive (DSA) LA algorithm proposed in this Letter is designed to reduce the transmission time of a video-frame. Hence, DSA seeks to minimise δ when $\delta > 0$ and to maximise $|\delta|$ when $\delta < 0$. Since DSA also aims at maximising the throughput, the algorithm tries, for each CS, to maximise (3a) if $\delta > 0$ or (3b) if $\delta < 0$:

$$\frac{Throughput}{\delta} = \frac{Throughput \cdot R_{CS-i}}{S - D \cdot R_{CS-i}} \tag{3a}$$

$$Throughput \cdot |\delta| = Throughput \cdot \left(D - \frac{S}{R_{CS-i}} \right) \tag{3b}$$

with the throughput as defined in (1).

For each video-frame and every CS, DSA computes δ. Depending on whether δ is positive or negative, (3a) or (3b) is evaluated for each CS and the optimum CS is selected according to the channel quality conditions.

Results: This study has been conducted using a burst-level event-driven simulator [3], with the main simulation parameters summarised in Table 1. Although this study concentrates on H.263 transmissions with a 16 kbit/s bit rate, two best-effort services, email and WWW, have also been implemented to create a mixed traffic scenario. In the simulations, TA was always used for WWW and email. For all LA algorithms, the adaptation occurs every 60 ms.

Table 1: Simulation parameters

Parameter	Value
Cluster size/sectorisation/cell radius	4/120°/1 km
Channels per sector/channel allocation scheme	16/random
Traffic load (downlink and single slot)	H.263 video (6 users/sector), WWW (3 users/sector) and email (3 users/sector)
ARQ protocol	Only for WWW and email users
Vehicular speed	50 km/h
Pathloss/shadowing	Okumura-Hata/log-normal distribution (6 dB standard deviation and a 20 m decorrelation distance)

An important benefit obtained with DSA is that its use guarantees a fairer operation of LA compared to the reference LA algorithms. In fact, DSA improves the QoS for the users experiencing a worse service. The simulations conducted have shown that DSA increases the minimum throughput guaranteed for 95% of the samples by 8.3% compared to TA, 14.2% compared to EA and 22.6% compared to MEA. The improved fairness of DSA is not obtained at the expense of the mean performance or the performance of users that previously experienced the higher throughput levels. In fact, Fig. 1 shows that DSA improves the H.263 throughput performance for the whole range of bit rates. Moreover, DSA increases the mean throughput performance by 2.3% compared to TA, 4.3% compared to EA and 10.6% compared to MEA.

The throughput improvements observed with DSA are due to a more aggressive CS selection policy and a better operation of LA. With the TA approach, the system utilises the highest data rate scheme (CS4) 59.8% of the time. This value increases to 65.7% with the DSA. Since CS4 also provides the lowest error protection, the DSA operation results in a higher average BLER (0.122 compared to 0.112 for TA). The EA and MEA schemes are characterised by a conservative approach in terms of CS selection (e.g. MEA exhibits a 34% CS4 usage) that results in lower average BLER values (0.106 and 0.0635 for EA and MEA, respectively). Despite a poorer error performance, DSA still increases the percentage of blocks received with the optimal CS (71.2%) compared to TA (66%), EA (64.1%) and MEA (43.4%). Improving the operation of LA results in important reductions in the average number of CS changes per second compared to TA and EA (16% and 22%, respectively), and consequently on the signalling load associated with the use of LA.

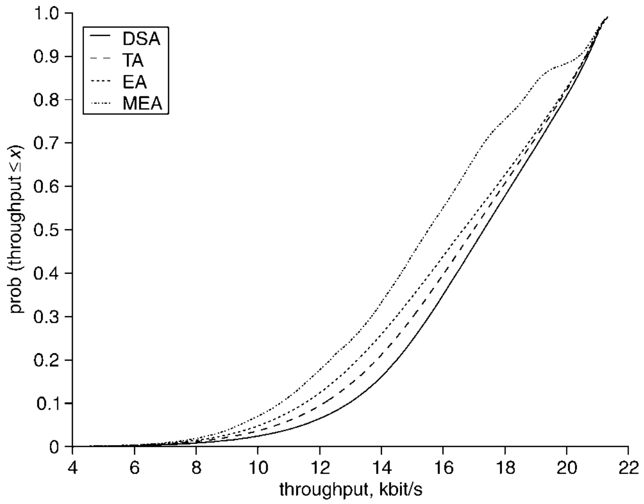

Fig. 1 *Cumulative distribution function of H.263 throughput performance*

The performance has also been evaluated in terms of the normalised delay, defined as the time needed to transmit a block of data divided by the size of such block. A significant reduction of this parameter has been observed with DSA. In terms of average normalised delay, the reduction is equal to 4.4%, improving the performance from 55.6 ms/kbit with TA to 53.2 ms/kbit with DSA; the reductions are much more important compared to EA and MEA (12% and 17%, respectively). In terms of the minimum performance guaranteed for 95% of the samples, the improvement obtained with DSA compared to TA increases to 18.8%, which results in the normalised delay being reduced from 86.3 to 70.1 ms/kbit.

The lower normalised delay obtained with DSA improves the real-time transmission of H.263. Table 2 shows that DSA achieves an important increase in the percentage of video-frames transmitted without delay, i.e. that their transmission is finished before the next frame has to be transmitted in real-time. In terms of video quality, [4] suggests an H.263 BLER target of 5% since no noticeable video degradation is produced below such target. Despite the higher average BLER observed with DSA, Table 2 demonstrates that DSA achieves a higher percentage of frames transmitted without delay and with the required target error rate.

Table 2: Results for real-time transmission of H.263 video frames (%)

	Without delay	With delay	Without delay and with BLER $\leq 5\%$
Throughput alg. (TA)	69.16	30.84	44.98
Error alg. (EA)	65.97	34.03	43.66
Modified error alg. (MEA)	49.71	50.29	39.94
Delay-sensitive alg. (DSA)	73.92	26.08	46.45

The best-effort services performance has also been analysed as part of this study. It is worth highlighting that using DSA for H.263 transmissions has not degraded the performance of the best-effort services.

Conclusions: A new LA algorithm designed to improve the transmission of services with tight delay constraints is presented. The proposed algorithm determines the optimum transport mode based on throughput and delay. The study has demonstrated that the proposed scheme is a suitable candidate for improving the real-time transmission of delay-sensitive services.

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