ALGORITHMS FOR LINK ADAPTATION IN GPRS

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Abstract - In the General Packet Radio Service (GPRS) development of GSM, four error correcting coding schemes have been defined to allow for variations in radio channel conditions. In this paper, two algorithms for link adaptation (i.e. choosing the most suitable coding scheme for transmission) are discussed. The first one is based on the estimated C/I (ECI) of the transmission and the second one on block error rate (BLER).

The results of the study indicate that algorithms based on estimated C/I are superior to those based on block error rate. The improvement in terms of throughput per user is in the order of 10%. Both algorithms are superior to any fixed coding scheme.

One problem that is apparent when using link adaptation is the degree of fairness in the system. Unfortunately, both the proposed algorithms show disturbing variations in throughput between different users. This is true also for the ideal case, where throughput is maximized by basing the choice of coding scheme on actual C/I.

I. INTRODUCTION

One of the developments of GSM (Global System for Mobile communications) is the GPRS (General Packet Radio Service) standard for packet radio transmissions. In GPRS four coding schemes (i.e. schemes for error correcting coding) have been defined to allow for different radio environments. The aim is to maximize the throughput of a channel by using the most suited coding scheme at each given moment. Simply put, a stronger coding scheme is more suitable for bad channels and a weaker coding scheme is more suitable for good channels.

The optimal choice of coding schemes can be described as a function of the carrier to interference ratio (C/I) of the channel, i.e. if the C/I is known a coding scheme that maximizes the throughput can be chosen (see Figure A). (5)

The problem is of course that in a real system, the C/I for a transmission that is being started is unknown and therefore has to be estimated. The issue is to identify an algorithm that uses some measurable parameter to approximate C/I and to choose the best coding scheme. To reach this goal, three questions have to be answered:

1. What parameters should be used to estimate the carrier to interference ratio?
2. How should the algorithms determining the choice of coding scheme be designed?
3. What performance measures should be used to evaluate different algorithms?

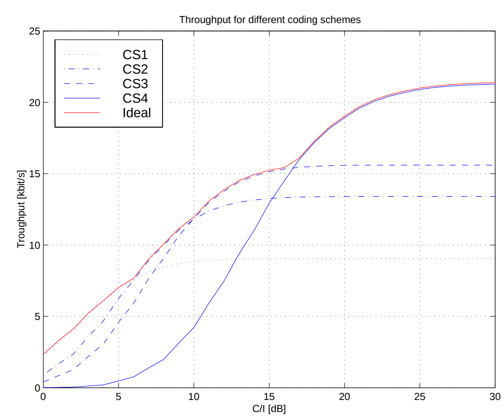

Figure A: Throughput vs. carrier to interference ratio.

GPRS basics

When data packets are transmitted over GPRS links they are split into smaller blocks of data. This is done in a hierarchical fashion, starting with the LLC frames. The packets are split into a number of LLC frames, each having a maximum length of 1600 bytes. The LLC frames are in turn split into RLC blocks. The number of data bits contained in each RLC block depends on the coding used.

One RLC block is finally transmitted on four consecutive bursts. (4)

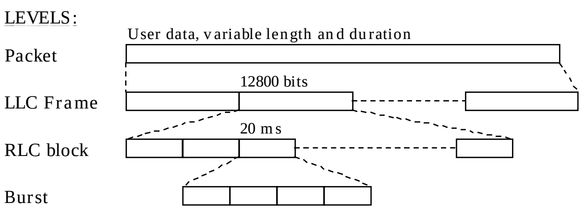

Figure B: Packet transformation data flow.

The GPRS standard specifies four different coding schemes, CS1 to CS4. The CS4 scheme does not use error correcting coding. CS1 uses a half-rate convolutional code. The CS2 and CS3 schemes use the same convolutional code, but with different degrees of puncturing. An RLC block can carry 181, 268, 312 or 428 data bits depending on the chosen coding scheme. (3)

For the downlink the transmission is initiated by the Basestation sending a paging message to the mobile. The mobile acknowledges the paging message and the base station starts its data transmission. Data is transmitted over one reporting window, which is a number of RLC blocks long. After each reporting window an ACK/NACK is expected from the receiver, followed by either further transmissions or retransmissions. (7)

II. PROPOSED ALGORITHMS

The algorithms proposed here take into account three aspects of the choice of coding schemes. First what coding scheme should be used initially, second how to update the chosen coding scheme, and third what system parameter to measure.

Initial coding scheme

There are several different approaches that can be used to determine the initial coding scheme to be used in the system. If the system has a relatively low load (typically below 50%) the simplest choice is to always start with the weakest coding, i.e. coding scheme 4. Most of the time the radio channels will have sufficient quality to allow this. Problems arise when the system load increases.

If we envision a system with moderate to high loads, the radio conditions will in general deteriorate due to increased interference. Here it becomes obvious that an initial choice of coding scheme 4 will lead to a large amount of retransmissions and at least one update of the coding scheme will be necessary to lower the block error rate to reasonable limits. It would therefore be beneficial to choose a coding scheme adapted to the system load.

As it turns out, the system load is known since the queues of traffic to and from mobiles are controlled by the base stations. The problem is however that the load varies through the system. What is of interest for each mobile or base about to initiate transmission is the current traffic from mobiles/base stations that interfere with them. The total system load is of lesser consequence.

In the algorithms presented in this paper the pragmatic approach has been used, i.e. coding scheme 4 is always chosen as a starting point regardless of system load.

Updating the coding scheme

The ideal solution for the updating of the coding scheme is known, with the optimum being a function of the carrier to interference ratio for the transmission. The C/I for a transmission is of course impossible to know beforehand, and can only be estimated (in a real system this is in fact true even for RLC blocks that have already been transmitted).

Two different methods for estimating the C/I of the transmission have been studied. In the first case a signal subspace approach was used to estimate C/I. In the second case, a coarser approach was used. Here the block error rate for a number of RLC blocks was used to indicate the C/I level and thereby change coding scheme.

<u>Estimated C/I algorithm:</u>

It has been shown that C/I can be estimated with a mean error of 1 dB in one burst time (2). In the system studied in this paper we estimate the mean C/I over the latest reporting window. This estimate is used to select the coding scheme providing the highest throughput for the given C/I (cf. Figure A).

Algorithms using an estimate of the average C/I should come fairly close to the ideal case. A good approximation of C/I has to be balanced against using large amounts of historical data. If the radio environment changes quickly, such averages become less useful. In the case studied here, with system loads around 50% and stationary mobiles, the changes are probably quite slow and the approximation very effective.

The estimated C/I algorithm implemented here can be described as:

1. Start with coding scheme 4.
2. Transmit RLC blocks until the end of the reporting window.

3. Estimate the average C/I for the previous reporting window. In the model this is done by averaging the obtained C/I for all RLC blocks. A random value is then added to the estimate to simulate uncertainties. The random value is lognormally distributed with a variance of 1 dB.
4. If there is more data to transmit return to step 2. Failed blocks will be placed at the head of the queue, and are retransmitted first in the next reporting window.

Block error rate algorithm:

For each reporting window the block error rate can easily be determined. However, due to the fact that the reporting window needs to be fairly short to allow for reasonably rapid updating of the coding scheme, the block error rate of an individual reporting window can only be determined as a coarse.

If the block error rate of the previous reporting window lies above a certain threshold, a stronger coding scheme should be used. If it instead lies below some other threshold, a weaker one should be chosen. In our simulations the upper threshold for BLER is 20% erroneous blocks and the lower threshold is 10% erroneous blocks.

The block error rate algorithm implemented here can be described as:

1. Start with coding scheme 4.
2. Transmit RLC blocks until the end of the reporting window.
3. Check how many RLC blocks that failed in the reporting window. If less than 10% of the RLC blocks failed change to a weaker coding scheme. If more than 20% of the RLC blocks failed change to a stronger coding scheme.
4. If there is more data to transmit return to step 2. Failed blocks will be placed at the head of the queue, and are retransmitted first in the next reporting window.

Coding schemes for retransmissions

When retransmissions occur in the system it is not always reasonable to change to a stronger code. If the block error rate is low it may still be relevant to keep a weaker coding scheme due to the benefit of a higher bit rate in the transmissions. Retransmissions in themselves, however, are always costly since they require a new ACK/NACK signaling between transmitter and receiver before further signaling is possible (i.e. some form of idle state). It can therefore be relevant to apply a stronger coding to the retransmissions than to the ordinary transmissions to avoid new retransmissions, even if the block error rate (or estimated C/I, whichever is being used) is below the threshold for changing schemes. In this study, the coding scheme used for retransmissions is determined by the behavior of the algorithms.

Packets that have not been transmitted successfully after 16 attempts are discarded. This is to prevent a mobile from occupying a channel indefinitely.

III. GPRS SYSTEM MODEL

The approach we have chosen is to make a numerical simulation of a GPRS system. The calculations have been made at the RLC block level. The reason for this is that the interference situation for a certain channel changes as new users start and stop their transmissions. This can happen from one RLC block to the next.

Model assumptions

In our simulations we have assumed the mobiles to be stationary. An adapted version of the RUNE tool for radio network simulation (1, 6) was used for the simulations. In this toolbox, calculations are made for standard (ideal) hexagonal cells. To allow for a finite number of cells to be simulated while still approximating an infinite system, a wrap-around technique is used. Only the downlink has been considered in this study.

The system can be described by the following parameters:

- Reuse pattern: 1/3.
- Cell radius 1000 m.
- 4 GPRS channels per cell.
- The packets arrive according to a Poisson process.
- The length of the packets is exponentially distributed with a mean of 600 bytes and a truncated maximum length of 68 Kbytes.
- The queuing discipline is FIFO.
- The system is loaded so that 50% of the channels are used for a fixed CS2 coding scheme. The load is constant for all simulations.

The propagation model includes distance attenuation where the signal power decays with the fourth power of the distance. It also includes a lognormal shadow fading with a lognormal variance of 8 dB.

Traffic

In the simulations a large number of mobiles are defined for each cell. For each mobile a state determining traffic status is defined for each time-step (equivalent to the length of an RLC block). There are four possible states:

1. Passive, i.e. not transmitting.
2. Queued, i.e. waiting to transmit.
3. Active, i.e. transmitting.
4. Silent, i.e. waiting for ACK/NACK.

Schematically the states and the changes between different states can be described by the diagram in Figure C. Note that the mobiles leave the passive state depending on the arrival of new traffic, which is Poisson distributed.

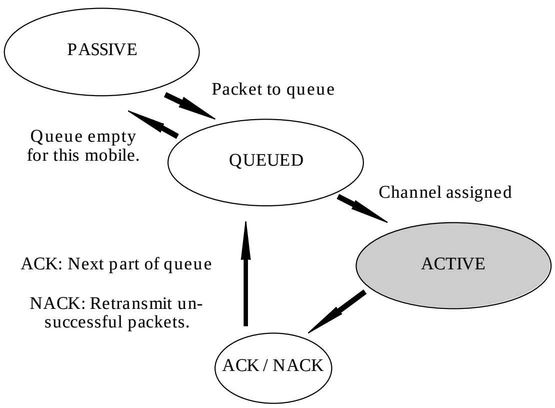

Figure C: Mobile terminal states during transmission.

The base station is assumed to control the queue. This means that the base station decides which mobile is allowed to transmit/receive at each given moment. In the simulation, there is therefore a queue for each cell where the identities of the mobiles wanting to transmit and the length of their respective transmissions are registered. Mobiles are randomly assigned a channel.

Transmission

When the mobiles are active, we assume that they transmit for example a maximum of 10 RLC blocks before waiting for an ACK/NACK. The length of this reporting window has been varied to see how this influences the system performance. Intuitively, in a given radio environment, a stronger coding scheme should work better with a longer reporting period since the risk of retransmissions being required is lower. Our studies indicate that long reporting windows are preferable, since this reduces the idle time during transmissions (i.e. overhead). However, long reporting windows mean that the coding schemes cannot be updated as often. Thus, in a quickly changing radio environment this may become a problem. In our calculations, a reporting window of 10 RLC blocks has been used.

A schematic description of our model of the transmission of packets on the channel can be found in Figure D.

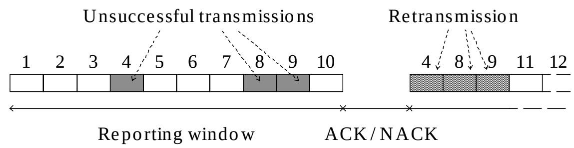

Figure D: Transmission of RLC blocks in one reporting window.

We simulate the ACK/NACK state by simply letting the mobile be silent for two RLC block windows (cf. 7).

The studied cases

Simulations have been made of three different groups of coding scheme choices. First of all, the performance for each of the individual coding schemes: CS1, CS2, CS3 and CS4, has been established. Second, the ideal case has been studied. Here the coding scheme has been selected using the C/I determined in the simulator for the RLC block that is about to be transmitted by each active mobile. Third, the two algorithms elaborated on above have been used to determine the choice of coding scheme.

In the algorithms proposed in this paper, the coding scheme of a certain transmission is updated only from one reporting window to the next. This means that there is always an idle state (cf. Figure D) between transmissions during which the ACK/NACK information is transmitted. The reason for this updating interval is that information on the quality of the received signal must be communicated to the transmitter in order for it to be able to evaluate the transmission parameters. Thus we assume that updates are not possible from one RLC block to the next.

IV. RESULTS

The foremost performance measure for a GPRS system if the possible throughput in bits/s. This has therefore been a natural starting point for the evaluation of the proposed algorithms. The ideal case and the strongest and weakest coding schemes respectively, have been our points of references (i.e. to which our results have been compared).

An uncritical use of only the throughput measurement, however, is not entirely satisfactory when judging the merits of different algorithms. Another important factor is the degree of fairness. Can different users expect approximately the same throughput or is the variation large? The variation in throughput over the users of the system is therefore our second performance measure.

In this study the discussion regarding different performance measures has not been carried further than this, although it is recognized to be an important issue. The results of our study are therefore presented as a cumulative distribution function of user throughput, see Figure E.

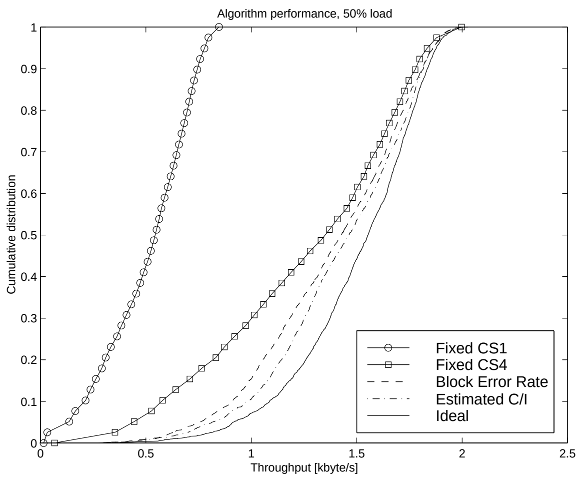

Figure E: The cumulative distribution function of user throughput for different linked adaptation algorithms.

As can be expected, the estimated C/I algorithm is superior to the block error rate in terms of user throughput. The gain that can be expected when using this more detailed measurement is in the range of 10% higher throughput. The throughput for the two algorithms only conforms for those users with a very high or very low throughput. This is due to the fact that in those cases no choice of coding scheme is made, the weakest one or the strongest one respectively is used continuously.

It is also apparent that both the proposed algorithms are superior solutions to any fixed coding scheme, but naturally inferior to the ideal solution (where the coding scheme would be adapted for each individual RLC block - before it was transmitted). Their performance is approximately 80-90% of that of the ideal solution.

Our conclusion is that there are definite benefits to be gained from implementing link adaptation algorithms in GPRS.

From the graph in Figure E we can see a potential problem with the implementation of GPRS. When link adaptation is used there is a large variation of throughput for the different users in the system. This means that although a user on average receives a high throughput, there is a fairly large risk of having a very bad channel and thus fairly low performance. This risk is alleviated if the system load is lowered.

Could the algorithms be designed to give a more fair performance? In a theoretical system this could be achieved by manipulating the queue for scheduled transmissions. By promoting users with bad channel conditions, a more evenly distributed throughput would be achieved (the user with good channel conditions would simply get longer queuing delays). The cost of doing this would of course be a lower average throughput. It would also be difficult to implement in a real system, since the future channel conditions for a user are hard to estimate before some transmission has taken place.

For a given fixed coding scheme, a very fair throughput spread is reached when the C/I for a large amount of the users is so large that very few retransmissions occur (this can easily be seen in Figure A). The necessary C/I levels vary depending on the coding scheme, for weak coding it is high and for strong coding it is low.

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