Adaptive Modulation for Fading Channels

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ABSTRACT

In future packet based wireless communication systems, transmission in the downlink will often dominate the traffic load. High bit-rate applications like WWW-browsing, file transfer, and full motion video will impose strong requirements on the system capacity. An obstacle in this context is the time-variability of the channel: For mobile users, frequently occuring fading dips will cause unnecessary, and capacity degrading, retransmissions.

To achieve a high throughput also over fading channels, adaptive methods for adjustment of e.g. the modulation alphabet, and the coding complexity, can be used. The idea is to make efficient use of the bits: Whenever channel conditions are adequate, transmission of redundant bits should be avoided.

In this paper we shall investigate the effect of adaptive modulation in a scenario involving one mobile and one base station.

INTRODUCTION

Fading channels confront us with the problem of lost packets and the need for frequent retransmissions. One strategy to combat time-variability is to use averaging: Spread-spectrum signalling can average out variations of the noise and interference level, while coding and interleaving can compensate for the temporary loss of signal strength due to fading dips. Such startegies can combat bad signalling conditions, but are inefficient when conditions are good. In the present PCC workpackage we explore a complement to the averaging strategy, where the time-variations of the channel, due to short-term fading, are estimated and the signalling scheme is adapted accordingly. We can exploit temporarily good transmission conditions to obtain higher throughput, while reducing the demands on the channel when its condition is bad. Assuming a system making use of either Frequency Division Duplex (FDD) or Time Division Duplex (TDD), with separate (ideal) control channels, the current channel parameters can be estimated and predictions about their future evolutions can be stored for subsequent transmission in the control channel. The bit-rate can be tailored to the current channel conditions by e.g. adjusting the modulation complexity, while keeping the transmitted symbol energy at a constant level. The further into the future the terminal can perform accurate predictions of the channel parameters, the more flexible and efficient the selection of the modulation alphabet will be. Moreover, the traffic on the control channel can be efficiently planned to minimize the signalling overhead.

For a predicted value of the signal to noise ratio (SNR) of the channel, the modulation level is maximized under the constraint of a certain probability of symbol error, for example, $P_M \leq 10^{-5}$. If no modulation level attains the required probability of symbol error, then transmission is deferred until later when the SNR is higher, thus avoiding retransmissions. The reason for using this strategy is that it will stabilize the error probability, thus keeping the retransmission rate at a low and constant level. The averaging strategies mentioned above do not have this feature. On the contrary, they would yield a higher traffic load when conditions are bad, since the increasing error rate would increase the requests for retransmissions.

Similar approaches have been proposed by Sampei, Goldsmith, and their co-writers in [1] and [2]. The main difference is our assumption that accurate long-term predictions of the channel parameters can be obtained.

SYSTEM DESCRIPTION

We shall outline and investigate a system that exploits the time-variations in the channel, instead of fighting their effects on the data being transmitted. The system presented in this investigation is intended to demonstrate the achieveable performance gains when using an adaptive approach to the problem of digital transmission over time-varying channels. In this investigation we use $64QAM$ as the maximum modulation level, thus transmitting six bits per symbol when the channel is as its best. When the channel degrades, lower powers of two are used with BPSK being the lowest level. In Figure 1 the left hand part illustrates the SNR-variation of a typical channel while the right hand part illustrates how the level of modulation can be selected for a pre-specified symbol error probability. As an example we note that for an SNR $\geq 20dB$ we can transmit during 18 time-units (Time $= 22$ to Time $= 40$) with a modulation level of $16QAM$ at a symbol error probability of $P_M \leq 10^{-5}$.

Slow power control is assumed to compensate for long-term fading, thus holding the long-term average of the received power at a constant level.

We have investigated two variants of the system: A TDD system for assymetric traffic, and an FDD system

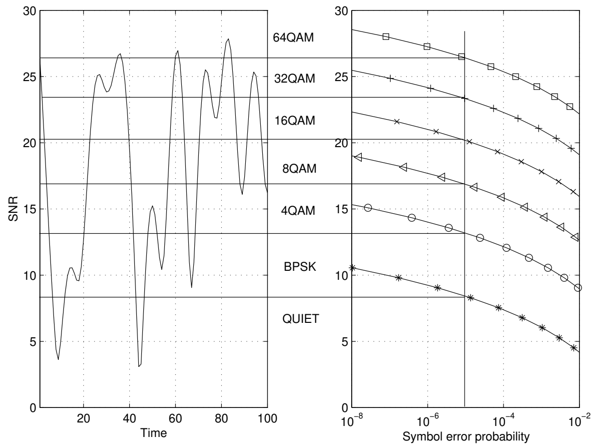

Figure 1: SNR profile and modulation level related to the error probability.

for symmetric traffic.

TDD system for asymmetric traffic

Time Division Duplex is a strategy for synchronising uplink and downlink transmissions. The base station and the mobile share a common frequency. An interesting feature of such a strategy is that if the uplink and downlink transmissions are closely spaced in time, it can be assumed that the channel conditions are correlated for the two transmissions. Under the assymetry assumption, the dominant part of the traffic will be carried in the downlink channel.

The current downlink SNR is then estimated in the mobile and fed into its predictor. The predictions are performed regularly, so that the mobile is able to make a decision on which modulation level to prefer for each SNR value. This decision only involves a table lookup, which is matching the SNR to the required BER. The mobile's decisions are collected in a buffer before transmitting its contents to the base station via the control uplink channel. The final decision on the choice of the downlink modulation is made by the base station, which signals its decision to the mobile, which in turn prepares for reception accordingly.

Since we use a TDD system, the downlink channel can be estimated through analysis of the uplink transmission, but since we assume an assymmetric traffic, this would require frequent transmission of pilot signals in the uplink. We therefore propose that that the estimator and predictor of the downlink quality should be placed in the mobile in this type of system. Although the traffic in the uplink doesn't require much transmission time, its quality needs to be guaranteed. For this reason the uplink modulation is also adapted to the channel conditions. The total system is schematically described in Figure 2.

FDD system for symmetric traffic

A different scenario would be that of a high, symmetric, traffic load. This would require equal resources in both

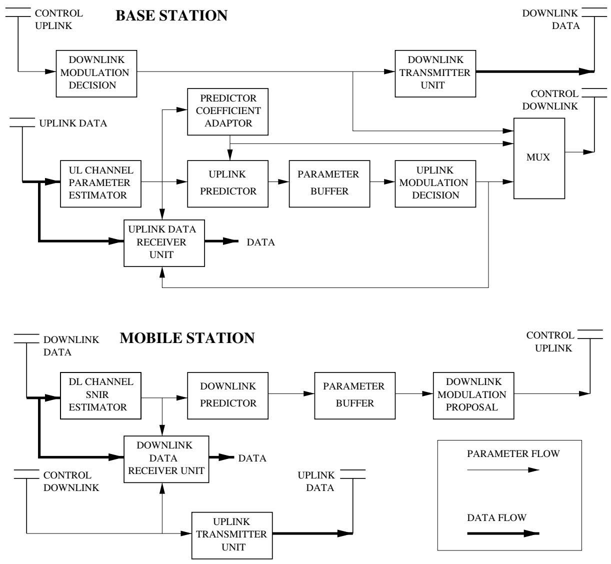

Figure 2: TDD downlink dominant system overview.

the uplink and the downlink. A natural way to accomplish such symmetric resource distribution is throudh Frequency Division Duplex, a counterpart to the TDD system described previously. In FDD, the uplink and downlink channels are independent, acting on separate frequencies, so they can transmit simultaneously without affecting each other. In this case, we have two different channels. Their characteristics need to be estimated and predicted to make the adaptive modulation work efficiently.

The complexity and frequency of the predictor parameter updating is a topic for further research, but it would probably require considerable computations. For this reason, the adaptor is proposed to be located at the base station.

For the case of FDD, depicted in Figure 3, the predictor has also been moved to the base station, since there is a potential advantage in providing the base station with the downlink SNR information: If the SNR information is known to the base station, it can control the different connections more efficiently. The price to be paid for this is an increase in traffic on the control uplink channel due to the necessity of transferring the estimated downlink SNRs from the mobile's estimator to the base station's predictor.

Thus the base station decides on both the uplink and downlink modulation formats, and signals the result to the mobile over the control downlink channel.

EXPERIMENT

To evaluate the proposed system solutions, a simulation series was conducted, assuming one base station transmitting to a mobile terminal. This is, of course, a simplification which implies that there will be no interference from other users. We also assume that the channel conditions are known for a few milliseconds ahead in time. Moreover, perfect synchronisation and transmission at a constant maximum amplitude, regardless of the modula-

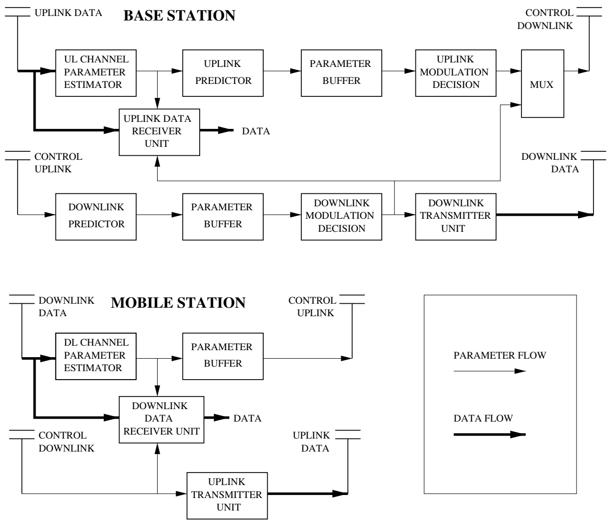

Figure 3: FDD symmetric system overview.

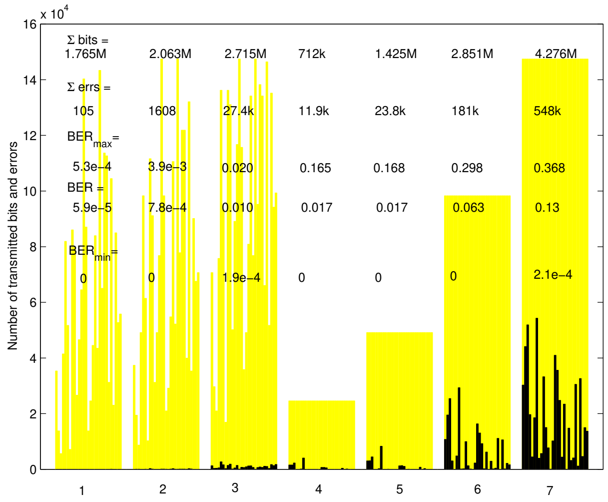

Figure 4: Transmission performance for three transmissions using adaptive modulation with different error-probability thresholds for the choice of modulation alphabet, and for comparison, four transmissions using constant modulation alphabets. BER_{max} and BER_{min} refer to the worst and the best frames in each transmission, whereas BER is the average bit error rate.

tion alphabet, is assumed. The experiment is applicable to both TDD and FDD systems, provided that accurate predictions of the channel conditions exist.

For each prediction of the SNR at the receiver, the modulation alphabet is selected for 512 consecutive symbols. This implies that the channel estimator and the predictor work at a rate of $\frac{bw}{512}$, where bw is the channel bandwidth. The data bit stream is then modulated and transmitted over the noisy channel. White Gaussian Noise (AWGN) with varying variance is added to simulate good and bad channel conditions. At the receiver, the signal is demodulated and the obtained bit stream is compared to the original one. The number of errors is counted, as well as the number of transmitted bits.

In Figure 4 the outcome of seven simulations is depicted: The first three columns belong to three transmissions using adaptive modulation with different error probability thresholds. The other four columns belong to four transmissions using constant modulations, namely BPSK, 4QAM, 16QAM, and 64QAM. The columns are subdivided into 29 subcolumns, each representing a frame of 48 time-slots of 512 symbols each. The light-gray columns illustrate the number of transmitted bits, whereas the black columns illustrate the corresponding number of errors. The accumulated number of bits for each transmission is written on top of each column, along with some statistics reflecting the bit-error rate:

BER_{max} The worst frame's bit error rate in that transmission.

BER The average bit error rate in that transmission.

BER_{min} The best frame's bit error rate.

As expected, the adaptive modulation approach results in a relatively constant (adjustable) error rate. On the other hand, the use of non-adaptive modulation results in high peaks in the error rate when the receiver encounters a fading dip.

The simulations were carried out using the IT++ package developed at the Communications Group, Department of Signals and Systems, Chalmers University of Technology.

CONCLUSIONS AND FUTURE WORK

The adaptive modulation approach provides a relatively constant error rate, which in turn provides an excellent basis for Forward Error Correction (FEC) codes, such as convolutional codes or block codes. For the non-adaptive case, clearly, the error rate peaks when the receiver enters a fading dip. This behaviour can probably not be compensated for by FECs, unless very large interleavers are used to average out the errors over time. The remaining errors after the decoding process will result in retransmissions, invoked by higher layers in the communication system. Obviously, such transmissions will increase the traffic over the channel.

By introducing the adaptive modulation approach, we gain two things:

1. The error rate is kept at a constant level, thus feeding the FEC algorithms with manageable data.

2. Radio transmission is postponed when channel conditions are bad, thus reducing the interference caused by other terminals.

This presentation only covers a single base station single mobile scenario. More general conditions need to be investigated before general conclusions can be drawn.

The following topics will be investigated in the immediate future:

- Multiple-access methods, using adaptive modulation under the assumption that channel parameters can be accurately predicted.
- The performance gains as function of prediction error levels and prediction horizon will be quantified.
- A deeper analysis of the required signalling overhead, the predictor initialization procedure, and the required hardware, will be carried out and included in the evaluations of the proposed systems.

REFERENCES

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[2] Soon-Ghee Chua and Andrea Goldsmith, "Variable-Rate Variable-Power MQAM for Fading Channels," *IEEE Vehicular Technology Conference Proceedings*, pp. 815–819, May 1996.