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Ulrike Korger / Christian Käs / Marko Hennhöfer / Martin Haardt

Cross-Layer Optimization of Streaming Media over Wireless MIMO Communication Systems

ABSTRACT

Ongoing developments in modern wireless multimedia applications require solutions that ensure almost constant **Q**uality **o**f **S**ervice (QoS) under the focus of minimizing the costs of transmission. Due to the rapidly changing channel conditions in wireless communications, inter-layer dependencies have to be taken into account leading to a so called cross-layer design. In this paper we present an approach which deals with the fact that the actual quality of time-critical streaming applications is a combination of three different parameters, namely the data rate of the stream, the maximum allowable bit error rate, and the tolerable delay per packet. Different combinations of these parameters can achieve the same QoS that is measured and represented by the *Peak Signal-to-Noise Ratio* (PSNR). To fulfill a given set of requirements, we combine trellis coded modulation with different transmission strategies on the physical layer. A simplified version of a Stop-and-wait **A**utomatic **R**epeat Re**q**uest (ARQ) system on the data link layer is taken into account as well. We derive analytical expressions for the transmit power and bandwidth consumption and calculate the overall costs of transmission. We finally optimize the system with respect to these costs. Thus the overall simulation time is rapidly decreased compared to solutions, where transmit power and bandwidth are stepwise increased and the optimum has to be chosen from a huge amount of possible combinations. Moreover, due to the analytical expressions for the costs, the exact requirements can be obtained, whereas in case of increasing the parameters stepwise an error dependent on the chosen step size occurs. Optimization takes place with respect to two different costs, namely the transmit power and the bandwidth consumption, and therefore a disproportional waste of one ressource is avoided.

1 INTRODUCTION

Streaming services are becoming more and more popular on the Internet. Ongoing developments in the field of wireless communications, like UMTS or HSDPA, will soon open up the mobile market to streaming media applications as well. Streaming media represents an application with both very high payloads and stringent QoS requirements, which makes it difficult to provide reliable and high quality media streams at a reasonable cost. This is the prerequisite for commercial distribution.

Traditionally, the optimization process in wireless communication systems is performed independently on each system layer (*intra-layer optimization*). In general, this approach does not result in an optimal set of parameters, as the inter-layer dependencies are neglected. Varying channel conditions on the air interface, challenging future multimedia services like mobile video conferencing, and the growing demand for QoS support in mobile environments necessitate the interworking between different system layers, leading to a *cross-layer optimization* approach. Parameters on different layers, which have the potential for optimization, have to be identified and properly chosen. In [1] a method is represented that uses equivalence classes of key-parameters of different layers and optimizes the system with respect to the transmit power costs. Different transmission strategies are compared. We combine this strategies with different modulation schemes and optimize the system with respect to two costs, namely the transmit power and the bandwidth consumption. Our approach deals with stringent time restrictions that occur in time critical multi media applications. A main focus for the optimization lies on the application layer, where QoS requirements are determined from a user point of view and on the physical layer, where these requirements have to be met in an optimal way. Here we use **T**rellis **C**oded **M**odulation (TCM) as an adaptive coding scheme [2], [3], which offers significant advantages compared to coding and modulation schemes that are separately chosen. The basic idea behind TCM is to choose subsets of the signal space in a way that allows the minimum Euclidian distance within these subsets to be maximized. Hence, different points within one subset are widely spaced and do not have to be coded. Therefore, this scheme offers coding gain without huge bandwidth costs. The network layer and the transport layer influence the optimization in terms of the chosen packet size and the protocol overhead.

The remainder of the paper is organized as follows: Section 2 describes the layer structure and the parameters, which combine the different layers for the optimization. The choice of different combinations of the three QoS requirements on the application layer, which ensures a certain state of QoS, is outlined in Section 3. Analytical expressions for the transmit power and the bandwidth costs to meet these requirements on the lower layers are presented in Section 4. Section 5 contains simulation results and Section 6 concludes the paper.

2 QoS REQUIREMENTS ON THE APPLICATION LAYER

In our proposed framework we optimize the system performance from a user point of view. To evaluate the Quality of the media stream different measures are available. To be more specific, we use the PEAQ (Perceptual Evaluation of Audio Quality) measure for audio streams and the PSNR for video streams. With this measures we have tools to define whether a stream appears good or bad from a user point of view. In the following, we explain a method to determine the parameter set that leads to the same QoS by means of a streaming video application. In a first step, a QoS-equivalence class has to be defined. With PSNR as the quality metric for video, different parameter sets that lead to the same PSNR have to be found. This QoS-equivalence class is described as a set of tuples t_i, with

$$T = \{t_1, t_2, ...\} \tag{1}$$

containing all possible combinations. The t_is are the interfacing parameters for cross-layer optimization, each containing a parameter triple, consisting of data rate *R*, a related maximal **P**acket **E**rror **R**ate (PER) and a service dependent maximum delay per packet Δ_p. Source distortion and packet loss distortion are the two contributing effects that result in a quality degradation of streaming video. In [4], an analytical model to determine the source distortion is developed. This distortion that is introduced by the encoding process mainly depends on the used codec, the bit rate, and the particular test sequence. The following formula is valid for H.264, a very common codec for video coding:

$$\mathrm{PSNR_S} = 10\log_{10} D_\mathrm{S}(R_\mathrm{S}) = a + b\sqrt{\frac{R_\mathrm{S}}{c}(1 - \frac{c}{R_\mathrm{S}})} \tag{2}$$

where $\mathrm{PSNR_S}$ is the source distortion in dB, D_S is the **M**ean **S**quare **E**rror (MSE) of the source distortion, R_S is the data rate and a, b, c are sequence dependent constants.
To describe the loss distortion, i.e., the video impairment due to lost packets, we use simulation results based on a publicly available toolset called EvalVid [5]. It turns out that the PSNR curve can be approximated by the help of an exponential function:

$$\mathrm{PSNR_{tot}} = \left(a + b\sqrt{\frac{R_\mathrm{S}}{c}(1 - \frac{c}{R_\mathrm{S}})}\right)\exp(-\lambda \cdot \mathrm{PER}) \tag{3}$$

Here λ is a parameter that has to be defined via measurements. With equation (3), QoS equi-

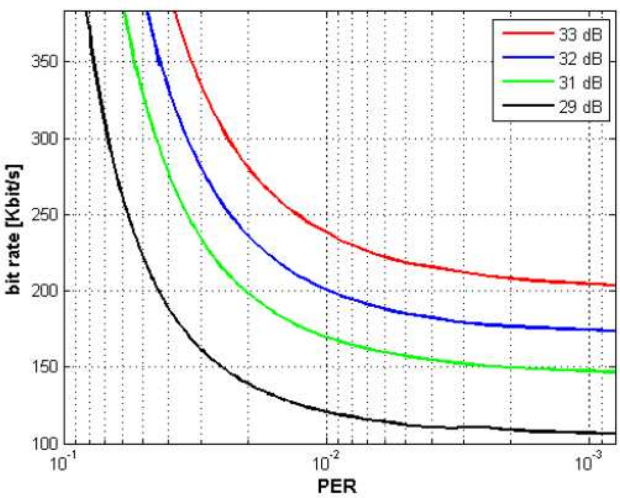

Figure 1: QoS equivalence classes

valence classes can be determined analytically with low computational effort, what makes the use of lookup tables dispensable. Figure 1 shows the resulting QoS equivalence classes. In the following, we describe the method to calculate the costs for different tripels of an equivalence class exemplarily for one tripel of QoS requirements.

3 LAYER STRUCTURE AND OPTIMIZATION PARAMETERS

We now explain how the different transport oriented layers of the ISO/OSI reference model affect the three main QoS requirements. Not every layer has a direct influence on every parameter. However, the choice of a certain strategy on one layer can change the demands of another layer. Figure 2 shows which layer influences which of the QoS requirements. A closer

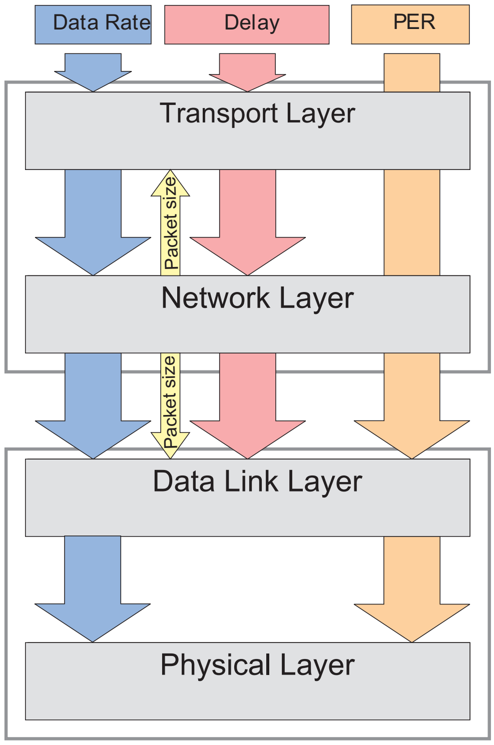

Figure 2: QoS requirements in the ISO/OSI reference model

look at the dependencies between the single layers can be found for example in [6]. Here we follow a ”Top-Down” approach, since we first examine the requirements of the top layer and then calculate the modified demands on the lower layers. Other approaches like [7] analyze the inter-layer dependencies starting at the lowest layer and are therefore called ”Bottom-Up” approaches.

The transport layer affects the requirements in terms of a protocol overhead. Therefore, the demands with respect to the data rate and delay increase for the lower layers. The network layer defines the size of the packets transferred over the channel. This layer has no direct effect on the QoS requirements, but determines the packet size, which, in turn, determines the requirements on the transport and data link layer. ARQ systems are implemented on the data link layer. In our approach we focus on a simplified version of a stop-and-wait ARQ system, which influences each of the requirements through the number of packet retransmissions. This number is between 0 and a maximum number of retransmissions that must not exceed the required delay time per packet.

We divide the transport-oriented layers into two units. One unit contains the transport layer and the network layer. The data link layer and the physical layer make up the second unit. We begin by identifying the influence of the first unit on the QoS requirements and then on making the second unit meet these modified requirements. The chosen packet size is communicated between the two units as well. This separation into two subunits allows us to circumvent the problem that the requirements for the physical layer and the data link layer cannot be examined separately because of their inter dependency.

4 ANALYTICAL EXPRESSIONS FOR THE TRANSMISSION COSTS

To find the ideal combination of the system parameters that minimize the costs of transmit power and bandwidth, these costs have to be expressed in terms of the constellation size M and the number of transmissions per packet n. This number equals 1 in case of no retransmissions and is equal to the number of retransmissions +1 otherwise. In this section we derive analytical expressions for the transmit power and bandwidth costs for the described trellis coded modulation with different MIMO transmission schemes. Afterwards we calculate the overall costs, find the minimum, and choose the corresponding combination of transmission strategy, constellation size and number of transmissions as the optimum solution.

4.1 Transmit power costs

To express the transmit power costs as a function of the constellation size and the number of transmissions, we follow a procedure shown in Figure 3. First, we calculate the **S**ymbol **E**rror

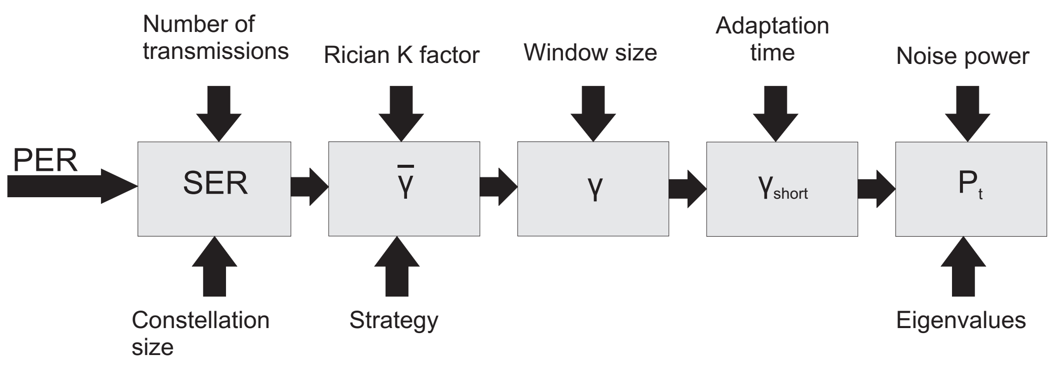

Figure 3: steps to calculate the transmit power costs

Probability (SER) which arises from the given PER requirement. In a next step, we evaluate the required mean SNR $\overline{\gamma}$ to meet this symbol error probability for every possible transmission strategy. This is done for every snapshot of the test scenario. We calculate the actual SNR

γ in a point within a certain window from the mean SNR of this window. As a last step we take into consideration that it takes some time, to adapt the transmitter and receiver to changing channel conditions. Therefore we hold the chosen system parameters constant over a certain period. For a reduced number of SNR values, referred to as γ_{short} in the figure, we calculate the transmit power requirements. Finally, we apply an analytical expression for the transmit power costs depending on the constellation size and the number of transmissions for all transmission strategies. The next parts give a more detailed description of the procedure.

4.1.1 Symbol error probability and mean SNR

Because in the described cross-layer design the required **F**rame **E**rror **R**ate (FER) and PER are constants, the required symbol error probability varies with the chosen constellation size and number of transmissions. In a first step, the symbol error probability is expressed dependent on the PER as follows:

$$P_{\mathrm{s}} = \left(1 - (1 - \mathrm{PER})^{\frac{L_{\mathrm{p}}}{\log_2(M)}}\right) \tag{4}$$

where L_{p} represents the packet size in bits including a header.
To consider ARQ on the data link layer, the number of transmissions has to be treated as a variable value. The maximal allowable number n_{max} is calculated dependent on the packet duration (t_{p}), the acceptable delay per packet (Δ_{p}) and the overall processing time per packet, also called round trip time (t_{RT})

$$n_{\mathrm{max}} = \left\lfloor \frac{\Delta_{\mathrm{p}}}{t_{\mathrm{p}} + t_{\mathrm{RT}}} \right\rfloor . \tag{5}$$

The symbol error probability after *n* transmissions is given by:

$$P_{\mathrm{s_{arq}}} = (P_{\mathrm{s}})^n . \tag{6}$$

To consider the constellation size, we express the required symbol error probability in terms of the squared normalized free distance ${\Delta_f}^2$. Afterwards, this distance is approximated through a 4-th order polynomal, called *poly(M)* in the following, which is a function of the chosen constellation size.

$$P_{\mathrm{s}} \approx 2Q\left(\sqrt{\frac{{\Delta_{\mathrm{f}}}^2 E_{\mathrm{S}}}{2N_0}}\right) \quad \mathrm{M-PSK} \tag{7}$$

$$P_{\mathrm{s}} \approx \frac{4(\sqrt{M}-1)}{\sqrt{M}} Q\left(\sqrt{\frac{{\Delta_{\mathrm{f}}}^2 E_{\mathrm{S}}}{2N_0}}\right) \quad \mathrm{M-QAM} \tag{8}$$

with:

$$\mathrm{Q}(\alpha) = \frac{1}{2\Pi} \int_{\alpha}^{\infty} \exp\left(\frac{x^2}{2}\right) \, dx .$$

The minimum normalized Euclidian distance for different code states and constellation sizes is listed in [8] for M-ary ASK, QAM, and PSK. The calculation of the normalized free distance (Δ_{f}) is straightforward.
In a next step we use the simulation data from the deterministic channel modeling tool IlmProp [9] that was developed at the Ilmenau University of Technology as basis for statistical upsampling. We assume a Rician distribution of the SNR in every temporal snapshot to approximate different realizations and achieve a mean SNR $\overline{\gamma}$. This is useful, if there is only one realization available, e.g., in case of measured data, and more generalized conclusions should be drawn. In a fading environment, $\overline{P_s}$ depends on the distribution of the fading amplitudes and therefore equations (7) and (8) change into:

$$\overline{P_{\mathrm{s}}} \approx \int_0^\infty 2Q\left(\sqrt{\frac{{\Delta_{\mathrm{f}}}^2\gamma}{2}}\right)p(\gamma)d\gamma \qquad \mathrm{M-PSK} \tag{9}$$

$$\overline{P_{\mathrm{s}}} \approx \int_0^\infty \frac{4(\sqrt{M}-1)}{\sqrt{M}}Q\left(\sqrt{\frac{{\Delta_{\mathrm{f}}}^2\gamma}{2}}\right)p(\gamma)d\gamma \qquad \mathrm{M-QAM.} \tag{10}$$

Here $p(\gamma)$ is the distribution of the SNR per symbol. Its distribution for the Rician channel can be expressed as follows [10]:

$$p(\gamma) = \frac{(1+K)e^{-K}}{\gamma}\exp\left(-\frac{(1+K)\gamma}{\gamma}\right)I_0\left(2\sqrt{K}\sqrt{\frac{(1+K)\gamma}{\gamma}}\right) \qquad \gamma \geq 0 \tag{11}$$

where I_0 is the modified Bessel function of the first kind and the 0-th order. Next the Chernoff-Bound is used as an approximation for the Q-Function:

$$Q(z) \leq \frac{1}{2}\exp(\frac{-z^2}{2}). \tag{12}$$

To evalulate expression (9) and (10), we employ the **M**oment **G**enerating **F**unction (MGF). For Rician fading $\overline{P_{\mathrm{s}}}$ can be calculated as follows:

$$\overline{P_{\mathrm{s}}} = a\frac{(1+K)}{1+K+\frac{\Delta_{\mathrm{f}}^2}{4}\overline{\gamma}}\exp\left(\frac{\frac{-K\Delta_{\mathrm{f}}^2}{4}\overline{\gamma}}{1+K+\frac{\Delta_{\mathrm{f}}^2}{4}\overline{\gamma}}\right) \tag{13}$$

where $a = 2$ for PSK and $a = \frac{4(\sqrt{M}-1)}{\sqrt{M}}$ for QAM.

4.1.2 Different transmission strategies

In the next section we focus on three different transmission strategies. Thereby we assume **C**hannel **S**tate **I**nformation (CSI) at the transmitter. The spatial modes are not interfering since we diagonalize the channel through SVD-based processing.
If the data is transmitted only over the strongest eigenmode, we call the transmission scheme

antenna mode (ANT), also known as dominant eigenmode transmission. In case of the ANT mode, the expression for the mean symbol error probability equals equation (13). This expression can be solved for $\overline{\gamma}$:

$$\overline{\gamma} = \frac{4 \cdot (K + K^2 - \lambda_W\{\frac{\overline{P_s} K}{a} \exp(K)\} - K \cdot \lambda_W\{\frac{\overline{P_s} K}{a} \exp(K))\}}{\lambda_W\{\frac{\overline{P_s} K}{a} \exp(K)\} \cdot \Delta_f^2}. \tag{14}$$

Here $\lambda_W(\cdot)$ is the Lambert-W function, which is the inverse function of $f(W) = W \exp(W)$. In the following we use the expression *diversity mode* (DIV), if the data is transmitted simultaneously over the two strongest eigenmodes and the transmit power is distributed on these two modes. Details of the power loading for the schemes are presented in subsequent sections. We assume **M**aximal **R**atio **C**ombining (MRC) at the receiver. In this case, the SNR of the single modes add. Unfortunately, for Rician fading no closed form expression for the sum distribution of the SNR per symbol after MRC is available. Therefore, the symbol error probability has to be expressed in terms of the SNRs per branch. We adapt the transmit power to the eigenvalues of the modes in that way, that every mode achieves the same SNR. Under the assumption of MRC, the symbol error probability is just the product of the MGFs associated with the SNR of each branch [11]:

$$\overline{P_s} = a \cdot \prod (M_{\gamma_i}(-b)). \tag{15}$$

Applied to the case of identical SNR on all modes, this leads to the following:

$$\overline{P_s} = a \cdot \frac{(1+K)^2}{\left(1 + K + \frac{\Delta_f^2}{4}\overline{\gamma}\right)^2} \exp\left(\frac{\frac{-2K\Delta_f^2}{4}\overline{\gamma}}{1 + K + \frac{\Delta_f^2}{4}\overline{\gamma}}\right). \tag{16}$$

This equation can be solved for $\overline{\gamma}$:

$$\overline{\gamma} = 4 \cdot \left(-\lambda_W\left(\frac{1}{\sqrt{\frac{a}{\overline{P_s}K^2}}} \exp K\right) + K\right) \frac{1+K}{\lambda_W\left(\frac{1}{\sqrt{\frac{a}{\overline{P_s}K^2}}} \exp K\right) \cdot \Delta_f^2}. \tag{17}$$

Because expression (16) contains the squared mean SNR, two solutions are available. The second solution leads to a negative mean SNR and thus negative power consumption and is therefore ignored.

A third transmission strategy divides the data stream into two sub streams, which are transmitted simultaneously over the two strongest eigenmodes. It is referred to as *multiplexing mode* (MUX) in the following. The overall symbol error probability approximately equals the sum of the symbol error probabilities of the single modes [12]:

$$\overline{P_s} \approx \overline{P_{s1}} + \overline{P_{s2}}. \tag{18}$$

If the content of a packet is distributed on the two modes, the overall transmission is only as good as the weakest mode. Therefore we again adapt the transmit power to achieve the same SNR on both modes. Using equation (13) for each mode, the overall symbol error probability can be expressed as follows:

$$\overline{P_{\rm s}} \approx 2a \cdot \frac{(1+K)}{1+K+\frac{\Delta_{\rm f}^2}{4}\overline{\gamma}} \exp\left(\frac{\frac{-K\Delta_{\rm f}^2}{4}\overline{\gamma}}{1+K+\frac{\Delta_{\rm f}^2}{4}\overline{\gamma}}\right). \tag{19}$$

This leads to the following expression for $\overline{\gamma}$:

$$\overline{\gamma} = 4 \cdot \left(K + K^2 - \lambda_W\left(\frac{\overline{P_{\rm s}}K\exp(K)}{2a}\right)\right) - \frac{K\lambda_W\left(\frac{\overline{P_{\rm s}}K\exp(K)}{2a}\right)}{\lambda_W\left(\frac{\overline{P_{\rm s}}K\exp(K)}{2a}\right)\cdot\Delta_{\rm f}^2}. \tag{20}$$

4.1.3 Actual SNR

Following the procedure depicted in Figure 3, we now calculate the actual SNR from the mean SNR. Therefore, as mentioned before, we compute the actual SNR within a window size *m* sample by sample from the mean SNR of this window. The window size depends on the coherence time of the channel. From:

$$\overline{\gamma(r)} = \frac{\gamma_{r-m+1} + ...\gamma_{r-1} + \gamma_r}{m} \qquad r = 1, ..., N : \text{actual sample number} \tag{21}$$

the actual SNR equals:

$$\gamma_r = m \cdot \overline{\gamma(r)} - (\gamma_{r-m+1} + ...\gamma_{r-1}). \tag{22}$$

Now we have the required actual SNR for every simulated temporal snapshot. To get the reduced series of actual SNRs, we calculate the maximal required actual SNR within the time that is needed for adaptation. Thus we assure that we do not violate the QoS requirements through the reduced time resolution of the adaptation process.

4.1.4 Transmit power

Finally, we calculate the transmit power required to achieve $\gamma_{\rm short}$. The actual SNR for the ANT mode can be calculated as follows:

$$\gamma_r = \frac{P_{\rm t}\sigma_1^2(r)}{N_0 B}. \tag{23}$$

Here, $\sigma_1^2(r)$ is the strongest eigenvalue of the *r*-th channel sample. Because we use simulation data from the IlmProp simulation tool, the exact value of the noise power density is known. A detailed description of the test scenario is given in the next section. To calculate the noise power density in case of measured data, we first have to estimate the power of the noise from the measurements. We assume constant AWGN (Additive White Gaussian Noise) over all dimensions (delay time, time, antennas, etc.). An effective way to measure the noise floor in the measurements is to observe the channel matrix in the time and delay time domain. If the path lengths

are short enough so that the last echoes extinguish before the maximum delay resolvable, we have a measurement of the noise without signal. If the number of samples is sufficient we can use this data to estimate the power of the noise. The calculation of the noise power density is straightforward.
Solving equation (23) for the transmit power and using equations (20) and (22) leads to an analytical expression for the required transmit power.
For the DIV mode and the MUX mode, we calculate the transmit power requirement separately for every mode:

$$P_{t1} = \frac{\gamma_r N_0 B}{\sigma_1^2(r)} \tag{24}$$

$$P_{t2} = \frac{\gamma_r N_0 B}{\sigma_2^2(r)}. \tag{25}$$

The overall transmit power consumption is the sum of the single transmit powers. Notice that the required bandwidth *B* appears,e. g., in equation (23). The calculation is explained in the following.

4.2 Bandwidth cost and cost weighting

To calculate the bandwidth consumption as a function of the number of transmissions and the constellation size, we use the following formula:

$$B = \frac{L_{\mathrm{P}} N_{\mathrm{I}}(\frac{1}{\mathrm{PER}} + (n-1))}{\frac{1}{\mathrm{PER}} T_{\mathrm{F}}(\log_2(M) - 1)} \tag{26}$$

where N_{I} is the number of packets per frame and T_{F} the frame duration. The order of the modulation scheme is reduced by 1, because we use trellis coded modulation with 1 bit for coding. Therefore, the bandwidth requirement reduces by the factor $\log_2(M) - 1$.
To calculate the overall costs of the different combinations we use a term, which we refer to as *weighted costs* C_{PB} in the following:

$$C_{\mathrm{PB}} = \left(\frac{P}{P_{\max}}\right)^{\alpha} \left(\frac{B}{B_{\max}}\right)^{1-\alpha}. \tag{27}$$

Here α is a user defined weighting constant between zero and one. This takes into account that different users have different amounts of both resources available and might therefore prefer spending more of one resource instead of the other. With the constant they can achieve their personal cost function structure. For scaling reasons we use the maximal useable bandwidth $B_{\max}$ and power $P_{\max}$ as reference. The expression is calculated for all constellation sizes and different numbers of transmissions. The optimum solution is the one with the lowest weighted costs.

5 SIMULATION RESULTS

In this section we show results for simulations with the IlmProp software tool. Figure 4 shows the used test scenario. The user and its trajectory are coloured in blue, whereas the position

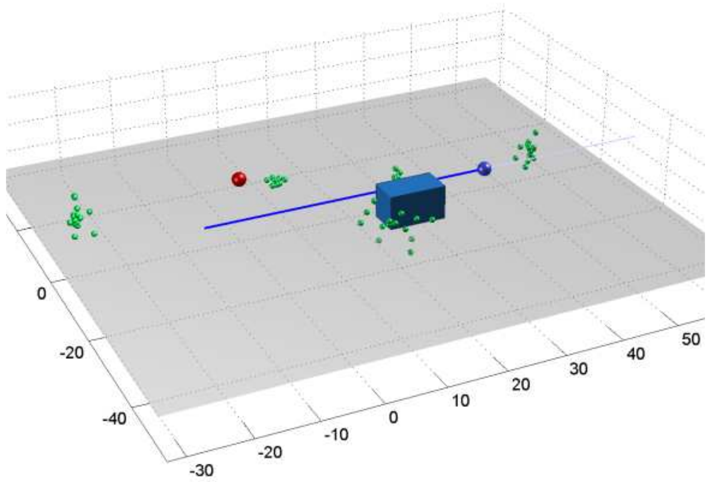

Figure 4: test scenario

of the base station is marked with a red circle. The blue box represents a building, which introduces shadow fading. Scatterers are depicted through green circles. We use 4000 temporal snapshots and assume a velocity of 160 km/h. The QoS requirements are set to a data rate of 512 kbps, a FER of 0.01 and a delay per frame of 100 ms. The assumed center frequency is 5 GHz. We restrict the available transmit power to 3 W and the maximal useable bandwidth to 500 kHz. Table 1 shows an overview over the possible system parameters for the test scenario.

For equally weighted transmission costs, presented in Figure 5, the DIV mode outperforms

TCM	8 PSK, 16 QAM, 32 QAM, 64 QAM, 128 QAM, 256 QAM
number of transmissions	1, ..., 3
transmission strategy	ANT, DIV, MUX

Table 1: overview over possible system parameters

the other modes at almost every time whereas the MUX mode causes the highest costs. This results from the fact that streaming media require very low error probabilities. The DIV mode achieves a diversity gain through transmitting the same information over both modes and is therefore optimal with respect to the error probability. Hence it achieves the required FERs with very low transmit power costs compared to the other schemes. The MUX mode, which achieves capacity gain and therefore is optimal with respect to high payloads, in turn suffers from errors on the second eigenmode, which results in an increased error probability. This can be observed, even if the bandwidth consumption is weighted higher, as depicted in Figure 6. Here, α equals 0.3. When shadowing occurs (sample 800-2000), the curve of the MUX mode is incomplete. Here, the QoS requirements cannot be fulfilled with this strategy due to the limited maximum available transmit power. Marker 1 shows an example for this situation. Notice that

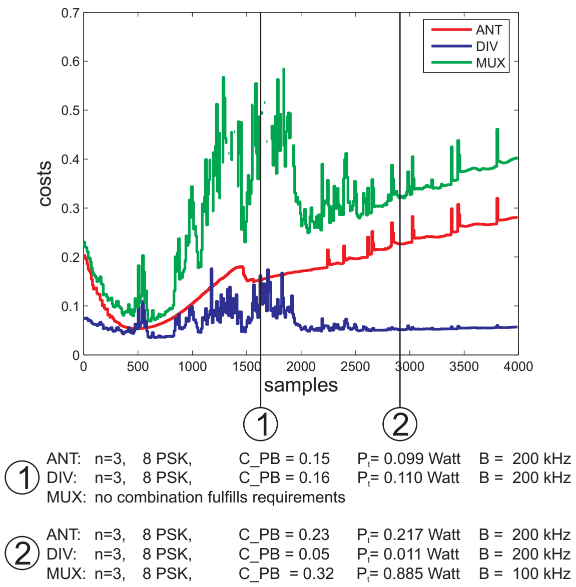

Figure 5: Costs for $\alpha = 0.5$

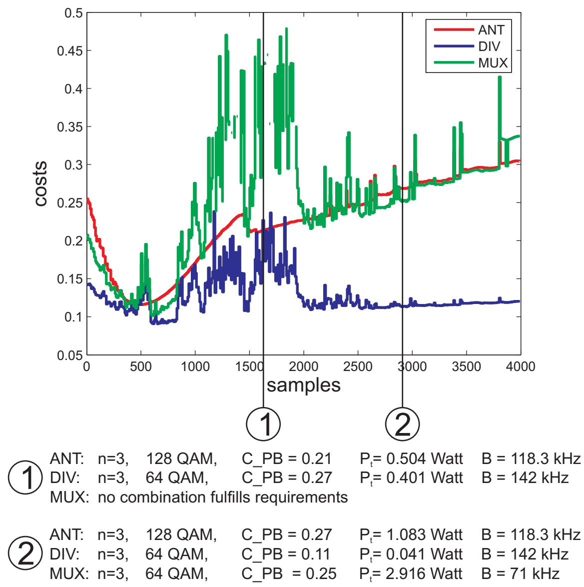

Figure 6: Costs for $\alpha = 0.3$

here the ANT mode even outperforms the DIV mode due to distortion, e.g., due to the fact that shadowing affects mainly the second eigenmode, which is not used in this scheme. Therefore this strategy can reach a higher constellation size, namely 128 QAM, compared to 64 QAM in the DIV mode case, hence saving bandwidth costs. Both strategies use the maximal number of packet transmissions, because the increase of bandwidth costs can be neglected compared to the savings of transmit power due to decreased error probability demands. Marker 2 shows an sample, where the second mode is not severely corrupted. Here, the MUX mode achieves lower costs than the ANT mode due to savings in bandwidth consumption, although it uses a lower modulation scheme. The DIV mode is optimal, because with increasing distance to the base station the costs to meet the error probabilities grow rapidly for the other two strategies whereas the curve for this mode rises slowly. Therefore the higher bandwidth costs are compensated.

6 CONCLUSIONS

In wireless communications, cross-layer optimization approaches should be used due to inter layer dependencies. For this optimization, specific parameters on the different layers have to be chosen and combined to minimize the overall power and bandwidth consumption. In our approach we derive methods to find possible combinations of three QoS requirements on the application layer that achieve the same PSNR, which is an objective measure for the actual quality of the stream. Following a top-down approach, we take into account the influence of intermediate layers of the ISO/OSI reference model on these requirements. We use TCM combined with ARQ to meet the demands. Thereby, we focus on three different transmission strategies. For the simulated scenario with moderate data rates but very high error probability demands, the DIV mode is almost always optimal. The ideal constellation size of the TCM varies for the different schemes. The optimal number of packet transmissions is equal to the maximum allowed, because the increased bandwidth is not significant compared to the fast diminishing transmit power consumption. We derive analytical expressions for the transmit power and bandwidth consumption and achieve the optimal combination of the different system parameters through these costs. This technique offers high advantages compared to a method where transmit power and bandwidth increase stepwise, and the optimal solution is chosen from all combinations that fulfill the required QoS. Due to the high number of different combinations in the latter case, the overall simulation time is rapidly decreased through our approach. Furthermore, we calculate the exact required transmit power and bandwidth, whereas in case of increasing the parameters stepwise an error dependent on the chosen step size cannot be avoided.

VI. ACKNOWLEDGMENT

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