Downlink Video Streaming for Users Non-Equidistant from Base Station

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Abstract. *We consider multiuser video transmission for users that are non-equidistantly positioned from base station. We propose a greedy algorithm for video streaming in a wireless system with capacity achieving channel coding, that implements the cross-layer principle by partially separating the physical and the application layer. In such a system the parameters at the physical layer are dependent on the packet length and the conditions in the wireless channel and the parameters at the application layer are dependent on the reduction of the expected distortion assuming no packet errors in the system. We also address the fairness in the multiuser video system with non-equidistantly positioned users. Our fairness algorithm is based on modified opportunistic round robin scheduling. We evaluate the performance of the proposed algorithms by simulating the transmission of H.264/AVC video signals in a TDMA wireless system.*

Keywords

Multiuser video streaming, non-equidistant users, cross-layer, outage probability, fairness, H.264/AVC.

1. Introduction

Wireless video streaming is one of the most popular services in the telecommunication industry. Even though video coding techniques evolve, the challenge that researchers in this area face remains the same: improve the video quality using fewer bits. Also the required features of the video remain the same: variable bit rate and low delay. The wireless transmission brings another set of adverse features to be dealt with, such as the varying channel gain and the scarcity of resources - bandwidth and power. The performance of the transmitted video in wireless environment can be improved using the cross-layer principle. Description of the cross-layer principle for video transmission can be found in [1]-[9]. Another possibility for performance improvement is to exploit the features of multiuser video transmission. Combination of the advantages of cross-layer principle and multiuser transmission is also possible. When optimizing the parameters from different layers in downlink multiuser video transmission two approaches can be taken based on the technique used to estimate the importance of the video sequences. In the first approach a model based rate distortion curve [10] is used. The drawback of this approach is that errors occur if the model is not accurate enough. The model is usually created for the entire video sequence, and, due to the differences in the content of different video frames, model mismatch is inevitable for some parts of the video sequence. The second approach is based on accurately calculating the importance of the video packets ([11]-[20]). In this approach three groups of algorithms can be differentiated. The first group of algorithms ([13], [15]) uses fixed low bit error rate for transmission of every packet, resulting in negligible packet error rate, and, thus, focuses on the scheduling procedure. This approach utilizes the available resources suboptimally. The second group of algorithms ([11], [18]) finds the parameters used for transmission based on optimization procedure that includes all the available options obtained by the abstraction layer (additional layer used to abstract important parameters from different layers), such as modulation, channel coding and TDMA (Time Division Multiple Access) scheduling scheme. This approach has high complexity which makes it unsuitable for practical implementation. The third group of algorithms represented by [17], focuses on resource allocation procedure that linearizes the rate distortion curve. This approach requires a non-convex optimization procedure. In cases where no resource constraints exist, the approach allocates all the available resources in one slot to a single user and never splits the video packets for transmission in different coherence intervals. Here, we propose a low complexity algorithm for multiuser video streaming where the parameters at the physical layer can be determined analytically.

Algorithms for multiuser video streaming differ depending on the communication system they are intended for. Algorithms in [11], [13], [15], and [18] are based on convolutional coding as the basic coding technique. Convolutional coding was the leading channel coding technique in the past decade, but with the advance of hardware and computational capabilities of communication devices, turbo codes and low density parity check codes are becoming mainstream coding techniques. These codes are used by the latest wireless communication systems, such as WiMAX, LTE, HSDPA. For wireless transmission with capacity achieving codes, according to [16], the outage probability of the transmission is the principal cause of errors. In this paper the wireless system is assumed to utilize capacity achieving codes.

The interest in multiuser video streaming has led to the development of several algorithms that can be used to improve the performance of multiuser video transmission in systems that use capacity achieving codes ([16], [17]), but resource allocation and the choice of physical layer parameters are still an open problem. Here we describe an analytical approach to the choice of parameters at physical layer for each user, based on greedy algorithm, which is the first contribution of this paper. Our solution is most effective for systems with users non-equidistant from base station because all users experience different channel conditions, and, therefore, use different optimal parameters at physical layer. The assumption of non-equidistant users is realistic in all practical wireless communication systems.

The second contribution of this paper comes from the proposed algorithm for QoS (Quality of Service) based fairness used for video streaming. The fairness issue is especially pronounced in systems where different users experience different average channel conditions. This is usually the case when users are at non-equal distances from the base station. Different fairness approaches combined with the opportunistic principle and their features are described in [22], [23], [24] but they do not refer to video transmission, so that some adjustments must be made for video transmission. Several papers address this issue, and described algorithms can be categorized in two basic categories: multiuser video transmission of scalable video, and multiuser video transmission of single layer video. The first category is addressed in [19] and because of the features of the scalable video, QoS based scheduling from [22] is chosen to achieve fairness. This algorithm uses long term statistical QoS, and it is suitable for scalable video transmission because it ensures that the basic layer for every video frame will be received. This is not the case in single layer video transmission where a sudden loss in the quality of the video frames is possible and can result in dissatisfaction of the user which can cause connection termination. Another algorithm for fairness in multiuser environment can be found in [14]. This approach is model based and includes source adaptation, but it does not consider packet errors and channel state variations during frame transmission. Our proposed procedure for resource allocation based on the Opportunistic Round Robin (ORR) scheduling principle from [23] and [24] is much simpler and can be used in rapidly varying environment.

In order to find the physical layer parameters for wireless multiuser video streaming, which is the first contribution of the paper, we utilize the packet ordering procedure, extensively used in the literature ([16], [17], and [25]). Based on the structure of the function for objective evaluation of the quality of the received video, for each packet of every user, we find parameters at the physical layer that consider the channel state and the video packet length, independently of the video content. Then, the importance of each packet of every user is calculated in zero error conditions for that packet. This approach can be used for users located at equal or nonequal distances from the base station. This algorithm is based on greedy framework. Its advantage comes from allowing the video packets to be sent in different coherence intervals and from splitting the video packets. Our theoretical results and simulations refer to a TDMA system with information about channel distribution available at the transmitter, but can be easily generalized to other multiple access techniques. The second contribution of this paper is the scheduling procedure for video transmission based on the modified ORR scheduling procedure that ensures fairness among users in every video frame. The algorithm chooses threshold for the quality of the video and when a user achieves it, the user is removed from the resource allocation procedure until all users achieve that threshold or complete their transmission.

The rest of this paper is organized as follows. In the second Section we describe the multiuser video system. In the third Section we explain the quality estimation in the transmission system and the packet ordering procedure. In Section four we present our first contribution, the procedure for finding parameters at physical layer. In the subsequent Section we describe our second contribution, the fairness aware scheduling algorithm based on the ORR scheduling paradigm. Simulation results are presented in Section six, followed by conclusions.

2. Downlink Multiuser Video System

We assume a downlink multiuser wireless system used for transmission of video signals to different users. The base station receives video signals from one or several video servers or users, in case of real time video streaming, allocates resources to different users and transmits the video to them. In the system considered here the video content is divided into video packets that are independently decodable (these video packets can be NAL (Network Abstraction Layer) units if H.264 is used). We assume that the video content can be sent from the video server to the base station error free and focus on the wireless environment. This approximation can be justified by the abundance of available bandwidth and power in the wireline domain. Each user receives transmission from the base station through a different wireless channel with a time varying channel gain. Some amount of information about the channel is sent through the feedback link to the base station. Depending on the speed of the channel state variation (coherence interval) and the feedback delay, this information can be used to adapt the parameters of the system to the channel state. The feedback link can additionally serve to send information about successful delivery of video packets back. This information can be used to facilitate the calculation of the importance of the video packets. The logical organization of the system is shown in Fig. 1.

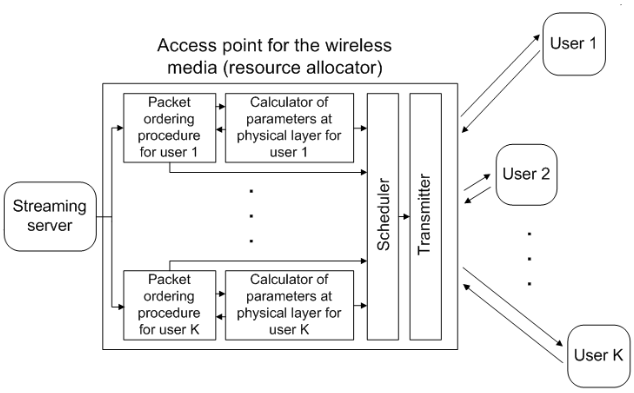


Fig. 1. Logical organization of the multiuser video system.

The base station consists of the following logical parts: packet ordering procedure, calculator of parameters at physical layer, scheduler and transmitter. The packet ordering procedure finds the importance of each video packet based on its video content, so that more important packets are sent and less important packets are discarded. The algorithm for importance calculation relies on finding the reduction of the distortion in the video signal that is caused by transmission of the packet with some error probability ([16], [17] and [25]). During the ordering process information from the calculator of parameters at physical layer is needed. The calculator of parameters at physical layer finds such parameters that result in the largest improvement of the quality of the received video of the respective user if resources are allocated to that user. In order to accomplish this goal the calculator must have some information about the channel of the respective user. The joint task of the packet ordering procedure and the calculator of physical layer parameters is to find the packet that causes largest distortion reduction and the physical layer parameters needed to obtain that reduction. This information is sent to the scheduler that allocates system resources to users in order to achieve largest improvement of the quality of the received video. In our analysis we consider a TDMA system, so that a specific time interval is allocated to each user. Then, the entire or part of the video packet from the user is sent from the logical block called transmitter to the respective user.

The maximal rate in number of bits per channel use for user k when capacity achieving codes are utilized, according to [21], is defined as:

$$c = \log_2(1+\nu\gamma_k) \tag{1}$$

where γ_k is the k-th user signal to noise ratio (SNR) and ν is a constant that is associated with the capacity achieving code parameters. In the rest of the paper we assume $\nu = 1$. The outage probability is defined as the probability that the utilized number of bits per channel use R_u is larger than the maximal i.e.

$$P_{out_k} = P(c = \log_2(1+\gamma_k) < R_u). \tag{2}$$

Different cases of channel information available at the transmitter (base station) can be used i.e. channel distribution information, imperfect or perfect channel state information. The parameters chosen at physical layer will be different for every specific case. If the delay of the feedback information is smaller than the channel coherence interval, the channel state information can be adjusted to the instantaneous feedback information and if the coherence interval is smaller than the delay of the feedback information, the parameters at the physical layer will be adjusted to the channel statistics obtained from all past feedback information.

As can be seen from Fig. 1, the users can be located at arbitrary distances from the base station so that they experience different average channel conditions.

3. Quality Estimation and Resource Allocation

As explained in the previous section, during the packet ordering procedure, information from the calculator of the parameters at physical layer is needed. The required information is the probability of error $p_{k,m}$ for the given video packet m of user k. We set this probability to be equal to outage probability $P_{out_{k,m}}$, where $P_{out_{k,m}}$ depends on the rate utilized for sending packet m of user k. Video packets contain coded bits for specific part of the video frame called slice, which in most cases is a row of macroblocks in a frame. According to [26]-[29], the expected distortion can be calculated as:

$$D = E\{(f_n^i - \tilde{f}_n^i)^2\} \tag{3}$$

where $\tilde{f}_n^i$ is the anticipated decoded value of the i^{th} pixel of the n^{th} video frame at the encoder, and f_n^i is the value of the i^{th} pixel of the n^{th} uncoded video frame. The expected distortion for packet m of user k can be calculated as:

$$D_{k,m} = (1-p_{k,m})D_{k,m_p} + p_{k,m}D_{k,m_n} \tag{4}$$

where D_{k,m_p} is the expected distortion with successful reception, and D_{k,m_n} is the expected distortion with unsuccessful reception of packet m of user k. The packet ordering procedure together with the calculator of the physical layer parameters, orders the packets according to their potential to reduce the expected distortion of the packet. Then the scheduler grants resources according to the following equation:

$$\min_{\{r_1,r_2,\ldots,r_K\}} \sum_{k=1}^{K}\sum_{m=1}^{M} D_{k,m} \tag{5}$$

where r_k stands for resources allocated to user k (these resources depend on the specific system - in the case of TDMA system r_k is the allocated number of symbols, in the case of CDMA (Code Division Multiple Access) system r_k is the allocated number of codes), K is the number of users in the system, M is the number of video packets used to encode a single video frame. The optimization of (5) is a very complicated task, especially since it is a non-convex optimization process (the expected distortion of a packet is generally neither convex nor concave function of the received resources). Another source of complexity is the potential necessity of splitting the video packets in several slots. Finally, the complexity is increased by the need to calculate the influence of the video packet on the future video frames as in [12] and the probability of sending the same video packet in future slots as in [25]. In order to make the solution feasible greedy based algorithms are utilized that make instantaneous solutions at every slot. Alternative form of (5) is presented in [17]. According to [17], the resources should be allocated in such a manner to cause largest reduction in the overall expected distortion:

$$\max_{\{r_1,r_2,\ldots,r_K\}} \sum_{k=1}^{K} -\frac{\partial E[D_k(p_k)]}{\partial R_k} R_k. \quad (6)$$

The variables R_k (the number of bits allocated to user k) and p_k are mutually related and depend on r_k. To solve (6) authors in [17] propose to use equal error probability for all packets sent by a user and linearize $-\frac{\partial E[D_k(p_k)]}{\partial R_k}$ over the limited number of bits that can be transmitted. According to these authors the following two stage procedure is optimal. The first stage consists of resource allocation to different users assuming a constant outage probability. After the allocated resources to user k are known, the optimal error probability is chosen based on 1D search. This second optimization is again non-convex and leads to solution that never splits video packets.

4. Choosing Parameters at Physical Layer

We denote the expected distortion when nothing is sent with D_{k,m_n} and the expected distortion when packet m of user k, containing $B_{k,m}$ bits, is sent under error probability $p_{k,m}$, with $D_{k,m}$. Using this notation the reduction of the expected distortion per bit for packet m can be calculated as follows:

$$\begin{aligned} -\frac{\partial E[D_{k,m}(p_{k,m})]}{\partial R_k} &= \frac{D_{k,m_n} - D_{k,m}}{B_{k,m}} \\ &= \frac{D_{k,m_n} - (1-p_{k,m})D_{k,m_p} - p_{k,m}D_{k,m_n}}{B_{k,m}} \\ &= (1-p_{k,m})\frac{D_{k,m_n} - D_{k,m_p}}{B_{k,m}}. \end{aligned} \quad (7)$$

From (7) and (6) we get:

$$\max_{\{r_1,r_2,\ldots,r_K\}} \sum_{k=1}^{K}\sum_{m=1}^{M} (1-p_{k,m})\frac{D_{k,m_n} - D_{k,m_p}}{B_{k,m}} R_{k,m}. \quad (8)$$

For TDMA system, (8) modifies as:

$$\max_{\{N_1,N_2,\ldots,N_K\}} \sum_{k=1}^{K}\sum_{m=1}^{M} (1-p_{k,m})\frac{D_{k,m_n} - D_{k,m_p}}{B_{k,m}} N_{k,m}R_{u,k,m} \quad (9)$$

where $N_{k,m}$ is the number of symbols allocated to user k for sending packet m and $R_{u,k,m}$ is the utilized rate per channel use, by the same user when sending packet m. The interdependencies among different packets from a single user are reflected in the values of D_{k,m_n} and D_{k,m_p}. Here we propose to use greedy algorithm that assumes that any packet has sufficient number of bits to utilize all available symbols in the given slot. Thus, in a TDMA system, the resources at any point of the algorithm will be allocated to the packet that has highest

$$(1-p_{k,m})\frac{D_{k,m_n} - D_{k,m_p}}{B_{k,m}} R_{u,k,m}. \quad (10)$$

The expression in (10) can be split into two mutually independent parts: $\frac{D_{k,m_n} - D_{k,m_p}}{B_{k,m}}$ which depends on the content of the video packet and $(1-p_{k,m})R_{u,k,m}$ which depends on the channel conditions and the length of the m^{th} video packet of user k. The first part can be treated as the importance of the video packet and it is constant at a specific moment, because the source coding has been carried out prior to the time when the transmission process takes place, and the expected distortion depends on the history of the transmission process. In order to maximize the reduction of the expected distortion, we maximize the term

$$(1-p_{k,m})R_{u,k,m}. \quad (11)$$

The solution for maximization of $(1-p_{k,m})R_{u,k,m}$ can be easily calculated for different types of channels. In order to account for cases when the video packet is sent in more than one coherence interval, we assume that the part of the video packet sent to a user in a single coherence interval is mapped to a single and independent MAC (Media Access Control) layer packet. If the video packet is split into two or more MAC layer packets, the parameters in (11) are the outage probability of the entire video packet and the average utilized rate per channel use. The calculation of the physical layer parameters, for the TDMA system with channel distribution information used in our simulation setup, is explained in detail in Appendix A.

It is necessary to point out the need of calculating D_{k,m_n} and D_{k,m_p} for every user after the transmission of each video packet, or after the reception of feedback information about the arrival of video packets from that user. As mentioned previously D_{k,m_n} and D_{k,m_p} are dependent on the known history of the transmission process i.e. the probability of error for different video packets that are ancestors of packet m, and transmission of a video packet or reception of feedback information changes the known history of the transmission process, so recalculation of D_{k,m_n} and D_{k,m_p} is needed.

Our algorithm for resource allocation is especially important in environments where users experience different mean channel SNRs, so that parameters at physical layer can be chosen for every user separately. Another benefit from our approach is the reduced computational complexity compared to both the optimization procedure using linearization and the optimization procedure where all the available options are considered.

The unequal error protection (UEP) principle, which has been shown to be very important, can be carried out in our algorithm by including already sent video packets in the packet ordering procedure for retransmission, but setting the importance of the video packet due to its video content to $\frac{(1-p_{k,m})(D_{k,m_n}-D_{k,m_p})}{B_{k,m}}$. To avoid the premature retransmission problem, in systems with variable delay, in the modified packet ordering procedure, the patient greedy algorithm from [25] can be used.

5. Fairness Algorithm

The optimization algorithm described in the previous sections is intended to minimize the overall distortion in the multiuser video system. In wireless environment, where users are located at different distances from the base station, the term described by (11) is generally larger for users located closer to the base station. If the difference in the values of (11) for different users is large enough, then the user with worse channel is expected to receive less system resources, which will lead to decrease of the overall quality of the user video and eventually to user dissatisfaction resulting in terminating the video streaming. Algorithms for fairness are divided into two major categories: long term and short term fairness algorithms. We argue that due to the nature of the video streaming, short term fairness algorithms have advantage over long term fairness algorithms when considering video transmission. The first shortcoming of the long term fairness algorithms is that they allow local performance minima which can have severe consequences on the user experience. Another shortcoming of the long term fairness algorithms is the long delay. Namely, delay restrictions imposed on video packets can be strictly met only by short term fairness (scheduling algorithms based on long term fairness can guarantee some level of service but not strict fulfilment of the delay requirement). For example, the authors in [30] claim that multiuser video transmission system based on proportional fair scheduling has problems with delay when video traffic comprises the majority of the overall traffic. The next reason for choosing short term fairness algorithms is based on our experience that the fairness parameters for all long term adaptation algorithms are strongly dependent on the specific video content that is transmitted, so the complexity of the short term resource allocation algorithms is much lower than the complexity of the long term algorithms.

Here we use a version of the opportunistic round robin scheduling (ORR) as the resource allocation procedure that guarantees fairness in the video quality of different users. Definition of classical ORR and a comparison to proportional fairness scheduling can be found in [23] and [24]. Our modification guarantees certain basic quality of the transmitted video for each user. Thus, we define a threshold level for the expected distortion D_b that should be achieved. At the beginning of the transmission of each video frame, resources are allocated according to (10) and all users are included in the resource allocation procedure. When a user experiences expected distortion lower then D_b he/she will be removed from the set of users that is included in the resource allocation procedure. If all users experience expected distortion lower than the previously agreed level then all users are again included in the packet ordering procedure.

D_b is defined by the required performance of the multiuser video system. According to [14] the average PSNR (Peak Signal to Noise Ratio) of 40 dB, 35 dB and 32 dB can be related to perfect, good and acceptable video quality described by mean opinion score measurement, respectively. Having those quality levels and the definition of PSNR, the threshold can be calculated as $D_b = \frac{255^2}{10^{\frac{PSNR}{10}}}$ and can be used in our proposed fairness algorithm. The benefits in the fairness obtained by our modified ORR will be demonstrated by simulation in the next section.

6. Simulation Results

In this section we present simulation results of the new algorithm of physical layer parameters calculation and of the resource allocation using the proposed modified ORR fairness scheduling algorithm.

6.1 Simulation Setup

We base our simulations on H.264/AVC JM v.16 video coder, publicly available at [31]. Five different QCIF (176x144) video sequences (foreman (user 1), news (user 2), hall monitor (user 3), mother and daughter (user 4) and carphone (user 5)) are encoded at a frame rate of 30 fps. Each video sequence is divided into slices, independently encoded. Every slice consists of a single row of macroblocks. Resynchronization markers are used in every slice. The first video frame is encoded as I frame and is available at the receiver with no transmission errors and all the other video frames are encoded as P video frames. In order to mitigate the effects of error propagation 15 macroblocks chosen randomly from every video frame are encoded intra (independently encoded) and the rest of them are encoded inter (using motion compensation). The video sequences are encoded using Content Adaptive Binary Arithmetic Coding (CABAC). For encoding each video sequence a constant number of quantization levels that produces a reconstructed video sequence of PSNR of 35 dB for 300 video frames of each video sequence, is used.

In our simulation scenario all users experience Rayleigh block fading channels with average channel SNRs of 20 dB, 23 dB, 26 dB, 29 dB and 30 dB, respectively. We assume that the transmitter has only statistical channel information in terms of the average channel SNRs of all users. The coherence interval of the channel is set to 1/6 of the duration of a single video frame transmission and is constant for all users.

At the receiver we use an error concealment method that uses the median motion vector from the macroblocks at the north-west (NW), north (N), and north-east (NE) position relative to the current macroblock. If the aforementioned motion vectors are not available, the zero motion vector is used for the error concealment procedure.

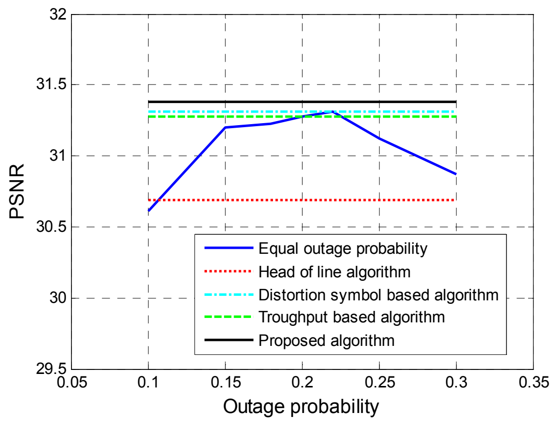


Fig. 2. The average PSNR using all proposed algorithms compared to the PSNR obtained by our proposed algorithm.

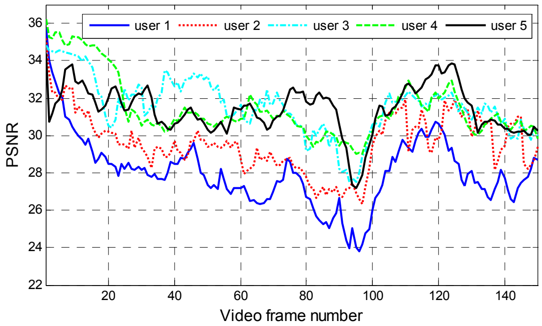
In order to calculate the expected distortion we use the ROPE algorithm from [26] and its distance adaptive correlation calculation, and quantization theory based rounding error compensation versions from [28]. The time after which the scheduler receives accurate information about the reception of video packets is set to two coherence intervals for all users. All simulation results are averaged over 20 different channel realizations.

6.2 Physical Layer Parameters

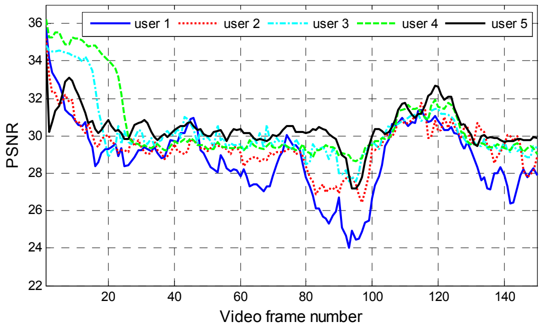
In order to evaluate the performance of our algorithm for resource allocation at physical layer, we use four different algorithms for comparison. In the first one different queue is created for every user and the user that has best linearized utility receives all the resources in a coherence interval. This is close to the algorithm in [17] and we call it head of line algorithm. We also use optimized version of this algorithm where distortion - TDM symbol curves are created for every user and those curves are used to allocate symbols to different users. This optimization is non-convex so near optimal solution based on Lagrangian optimization and subsequent greedy optimization is found. We call this algorithm distortion symbol algorithm. Another algorithm for comparison is based on system where all users use the same outage probability so that the number of used bits per symbol is calculated based on the outage probability. The last algorithm used for comparison is the same algorithm as the proposed one, but optimizes only the throughput in the system not considering the splitting of video packets in different coherence intervals. This algorithm is called throughput based algorithm. In all the systems above we used a symbol rate of 108 ksym/s. For the proposed algorithm the parameters at physical layer are calculated according to the method explained in Appendix A. The performance of the system with equal outage probability is shown in Fig. 2. In this figure, additional lines that describe the average PSNR for all the other algorithms are shown. The additional lines are not dependent on the outage probability but are shown for comparison reasons. Based on Fig. 2 several observations can be made. First, the performance of the algorithm based on linearized utility shows lower overall quality. This is due to the low number of coherence intervals available in the system, so that there are no enough transmission opportunities. The next observation is that the performance of the algorithm based on distortion symbol curves and the algorithm that does not consider packet splitting are very close. This is due to the fact that they both utilize the cross layer principle and are both aware of the influence of the user location, but the complexity of the first one is much higher due to the creation of the curves and the additional optimization. The improved performance of our proposed algorithm is due to considering the splitting of the packets that are sent in more than one coherence interval. This improvement is of order of 0.1 dB. The last observation is that the performance of the algorithms that use equal outage probability can be close to the performance of the other algorithms when the outage probability is found optimally. Also, the performance of the equal outage probability system is not very sensitive to the value of the error probability for a broad range of values.

6.3 Performance of the Modified ORR Algorithm for Resource Allocation

To evaluate the performance of the proposed modified ORR algorithm we carried out extensive simulations using different symbol rates. The PSNR threshold was set to 29 dB. Average PSNR of each user in terms of the video frame number is shown in Fig. 3 for a symbol rate of 90 ksym/s. The reduction of the PSNR variation of different users is obvious. User 1 whose PSNR is lower than the threshold PSNR has the largest gain that comes from utilizing resources previously used by other users. This effect in terms of the average PSNR of the whole video sequence of each user can be seen in Fig. 4. As shown, the difference between the average PSNR of the best and the worst user in the system without ORR is 3.6 dB, while in the system that uses ORR, this difference is reduced to 1.8 dB.



(**a**) Without modified ORR



(**b**) With modified ORR

Fig. 3. PSNR of all users in terms of video frame number.

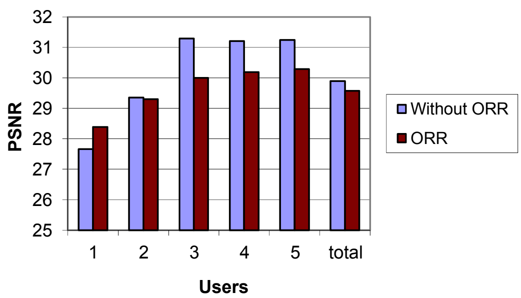


Fig. 4. Average PSNR for different users in the two systems for symbol rate 90 ksym/s.

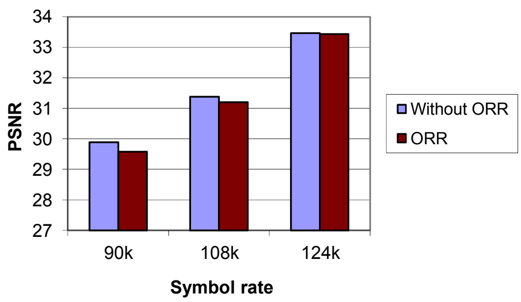


Fig. 5. The total average PSNR in the two systems for different symbol rates.

The difference in the total average PSNR reduces as the symbol rates exceed the ones from the region where the average PSNR of users is closer to the threshold level. This behavior can be seen in Fig. 5.

7. Conclusion

In this paper downlink multiuser video streaming for users non-equidistant from base station is considered. A new algorithm for allocation of resources is proposed which assumes that each packet can utilize the available resources and leads to partial separation of the physical and application layers. The algorithm shows improved performance compared to existing algorithms at lower or comparable complexity. The issue of fair resource allocation is also addressed and a new modified opportunistic round robin algorithm is proposed. This algorithm results in significant reduction of the PSNR variation for different users at modest or no degradation in the average system PSNR.

8. Appendix A

In our simulation scenario the system is considered to be TDMA and the user, chosen for transmission, is allocated as many symbols as needed to transmit the chosen video packet. The average channel SNR for the chosen user is $\bar{\gamma}$. If the required number of symbols is greater than the one available in the current coherence interval, then part of the symbols are transmitted in the next coherence interval. During the simulations we treated four different cases for the optimization process described in (11):

Case 1 No bits from the current packet are sent and all bits can be sent in a single coherence interval

Case 2 Part of the packet is already sent and the rest of the packet can be sent in a single coherence interval

Case 3 No bits from the current packet are sent and the bits must be sent in two different coherence intervals

Case 4 Part of the packet is already sent and the rest of the bits must be sent in two different coherence intervals.

The cases when three or more coherence intervals are needed to send the entire video packet are not treated because the number of available symbols in a single coherence interval is large enough so that this never happens. These cases can be treated as sub cases of cases 3 and 4. For our system the optimization in (11) was carried out using the following equations:

Case 1:

$$\max_{\gamma_0} e^{-\frac{\gamma_0}{\bar{\gamma}}} \log_2(1+\gamma_0).$$

This optimization can be carried out by solving the following equation:

$$ye^{y} = \bar{\gamma}$$

where $1+\gamma_0 = e^{y}$. γ_0 is the channel SNR used for transmission.

Case 2:

$$\max_{\gamma_0} e^{-\frac{\gamma_0}{\bar{\gamma}}} \frac{b}{N_1 + \frac{b - N_1 S}{\log_2(1+\gamma_0)}}.$$

This optimization can be carried out by solving the following equation:

$$ye^y((\tfrac{b}{N_1} - S)\ln 2 + y) = \bar{\gamma}(\tfrac{b}{N_1} - S)\ln 2$$

where $1+\gamma_0 = e^y$, b is the number of bits in the video packet, N_1 is the number of symbols used in the transmission of the previous parts of the video packet and S is the average utilized rate per channel use for the transmission of the previous parts of the video packet. γ_0 is the channel SNR used for transmission.

Case 3:

$$\max_{\gamma_0} e^{-\frac{\gamma_{01}+\gamma_{02}}{\bar{\gamma}}} \frac{b}{N_{a1} + \frac{b - N_{a1}\log_2(1+\gamma_{01})}{\log_2(1+\gamma_{02})}}.$$

This optimization can be carried out by solving the following equations:

$$\begin{aligned} e^x(\frac{b}{N_{a1}}\ln 2 + y - x) &= \bar{\gamma}, \\ ye^y(\frac{b}{N_{a1}}\ln 2 + y - x) &= \bar{\gamma}(\frac{b}{N_{a1}}\ln 2 - x) \end{aligned}$$

where $1+\gamma_{02} = e^y$, $1+\gamma_{01} = e^x$, b is the number of bits in the video packet, N_{a1} is the number of symbols available in the first coherence interval. γ_{01} and γ_{02} are the channel SNRs used to transmit in the first and in the second coherence interval.

Case 4:

$$\max_{\gamma_0} e^{-\frac{\gamma_{01}+\gamma_{02}}{\bar{\gamma}}} \frac{b}{N_1 + N_{a1} + \frac{b - N_1 S - N_{a1}\log_2(1+\gamma_{01})}{\log_2(1+\gamma_{02})}}.$$

This optimization can be carried out by solving the following equations:

$$\begin{aligned} e^x(\frac{b-N_1S}{N_{a1}+N_1}\ln 2 + y - \frac{N_{a1}}{N_{a1}+N_1}x) &= \frac{N_{a1}}{N_{a1}+N_1}\bar{\gamma}, \\ ye^y(\frac{b-N_1S}{N_{a1}+N_1}\ln 2 + y - \frac{N_{a1}}{N_{a1}+N_1}x) &= \\ \bar{\gamma}(\frac{b-N_1S}{N_{a1}+N_1}\ln 2 - \frac{N_{a1}}{N_{a1}+N_1}x) & \end{aligned}$$

where $1+\gamma_{02} = e^y$, $1+\gamma_{01} = e^x$, b is the number of bits in the video packet, N_{a1} is the number of symbols available in the first coherence interval, N_1 is the number of symbols used in the transmission of the previous parts of the video packet and S is the average utilized rate per channel use for the transmission of the previous parts of the video packet. γ_{01} and γ_{02} are the channel SNRs used to transmit in the first and in the second coherence interval.

The equations for all four cases were obtained by setting the first derivative of the optimized function to zero. For finding the solution of these equations we used numerical optimization based on the function fsolve from the Matlab Optimization Toolbox.

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