# Multiuser Cross-Layer Resource Allocation for Video Transmission over Wireless Networks

**Guan-Ming Su, Zhu Han, Min Wu, and K. J. Ray Liu, University of Maryland, College Park**

**Abstract**

With the advancement of video-compression technology and the wide deployment of wireless networks, there is an increasing demand for wireless video communication services, and many design challenges remain to be overcome. In this article, we discuss how to dynamically allocate resources according to the changing environments and requirements, so as to improve the overall system performance and ensure individual quality of service (QoS). Specifically, we consider two aspects with regard to design issues: cross-layer design, which jointly optimizes resource utilization from the physical layer to the application layer, and multiuser diversity, which explores source and channel heterogeneity for different users. We study how to efficiently transmit multiple video streams, encoded by current and future video codecs, over resource-limited wireless networks such as 3G/4G cellular system and future wireless local/metropolitan area networks (WLANs/WMANs).

Over the past few decades, wireless communications and networking have experienced unprecedented growth. With the advancements in video-coding technologies, transmitting real-time encoded video programs over wireless networks has become a promising service for such applications as video on demand and interactive video telephony. In most scenarios, multiple video programs are transmitted to multiple users simultaneously by sharing resource-limited wireless networks.

The challenges for transmitting multiple compressed video programs over wireless networks in real time involve several aspects. First, wireless channels are impaired by detrimental effects such as fading and co-channel interference (CCI). Second, radio resources such as bandwidth and power are very limited in wireless networks and should be shared among multiple users. In addition, unlike the transmission of generic data and voice, the rates of compressed video programs can be highly bursty due to the differences in video contents and intra/intercoding modes, which complicates source-coding rate allocation. Moreover, the optimizations in different layers are cross-related and for the most part have both continuous parameters and integer parameters, so that the formulated problem is often *NP* hard. Further, handling multiple video streams over a wireless system involves several important service objectives, such as system efficiency and individual fairness, and there are inherent trade-offs among these objectives.

To overcome the aforementioned design challenges, researchers often employ dynamic resource allocation, which allows for jointly adjusting the system's parameters and utilizing the limited system resources optimally in the source-coding and communication layers for multiple users. We focus on two major aspects to optimize resource allocation, namely, cross-layer design and multiuser diversity, for accommodating a large number of users with acceptable received quality of service (QoS). Specifically, in this article we discuss the multiuser video system in 3G/4G cellular networks and wireless local/metropolitan area networks (WLANs/WMANs) from a resource-allocation point of view. We first present a generic framework along with the wireless network components, the state-of-art video source codecs for a multiuser wireless video system, and the design principles of multiuser cross-layer resource allocation. We then investigate a few major design methodologies for multiuser video communications with cross-layer design. The goal is to provide researchers whose expertise is in individual layers with a broad overview of multiple relevant layers and with perspectives on the recent advances and future potentials in multiuser wireless video transmissions.

## Framework of Video over Wireless Networks

In this section, we first present an overview of video over wireless networks with multiuser cross-layer design. We then briefly review the communication and video codec subsystems, discuss the available resources and control parameters, and analyze the corresponding constraints in practical implementation. Finally, we study the design principles for multiuser cross-layer resource allocation.

### Overview

Figure 1 depicts a generic framework for multiuser video transmission over wireless networks, which consists of the video coding subsystem, communication subsystem, receiver subsystem, and resource allocator. The video coding subsystem encodes each incoming video frame in real time and sends the corresponding rate and distortion (R-D) information to the resource allocator. The communication subsystem analyzes the available resources in the network layer, medium access control (MAC) layer, and physical (PHY) layer, and supplies channel information feedback from the receivers. After gathering the information, the resource allocator executes optimization algorithms and allocates system resources to different links of different layers so as to achieve the net-

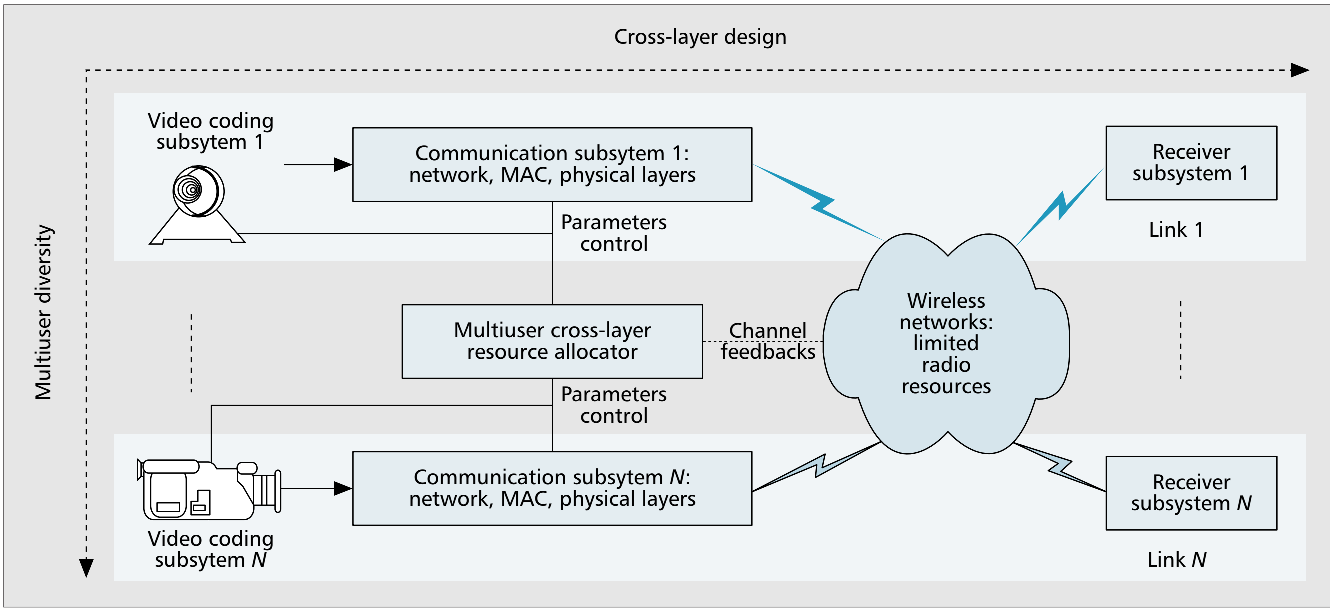



**Figure 1.** *Framework of a multiuser cross-layer video transmission system over wireless networks.*

work optimization objectives. We consider the following two design aspects:

• **Cross-layer Optimization**: Traditionally, wireless networks are designed in layers. Using Shannon's Separation Theorem, source and channel coding can be designed separately while still achieving optimality, if arbitrarily long delay is allowed. However, in most practical wireless networks, Shannon's Separation Theorem does not hold. One of the reasons is because the packets must be transmitted with delay constraints, especially for real-time video transmission. Subject to the current layered-design network, system designers often perform cross-layer optimization to achieve systemwide optimality.

• **Multiuser Diversity**: In a multiuser system, a limited amount of system resources are shared to transmit multiple video streams. The channel conditions experienced by different users are different and time-heterogeneous. In addition, the transmission rates of different video bit streams vary among users and over time. By exploring multiuser diversity in both source and communication subsystems and allocating system resources dynamically, we can improve the network performances and guarantee QoS satisfaction for individual users.

## Current and Future Wireless Network Paradigms

Subject to limited radio resources, modern wireless networks often adopt the following resource-allocation methods to adaptively improve spectrum utilization.

• **Adaptive Modulation and Coding (AMC)**: Adaptive modulation is an effective method to improve bandwidth efficiency. To combat different levels of channel errors, adaptive forward error coding (FEC) is widely used in wireless transceivers. Further, joint consideration of adaptive modulation and adaptive FEC provides each user with the ability to adjust the transmission rate and achieve the desired error-protection level, thus facilitating adaptation to various channel conditions.

• **Power Control**: The gains of wireless channels generally fluctuate over time. To maintain link quality, the signal-to-interference-and-noise ratio (SINR) should be dynamically controlled to meet a threshold known as the *minimum protection ratio*. This threshold depends on many factors such as the AMC rate and desired bit error rate (BER). The objective of power control is to guarantee certain link quality and reduce co-channel interferences.

• **Channel Assignment**: The channel used here is a general concept representing the smallest unit of radio resources that a user can be assigned for transmitting data, such as frequency band and time slot. Taking into consideration the different channel conditions and users' transmission requirements, dynamic channel assignment can improve the utilization of system resources by exploring the diversities over multiuser, time, and frequency.

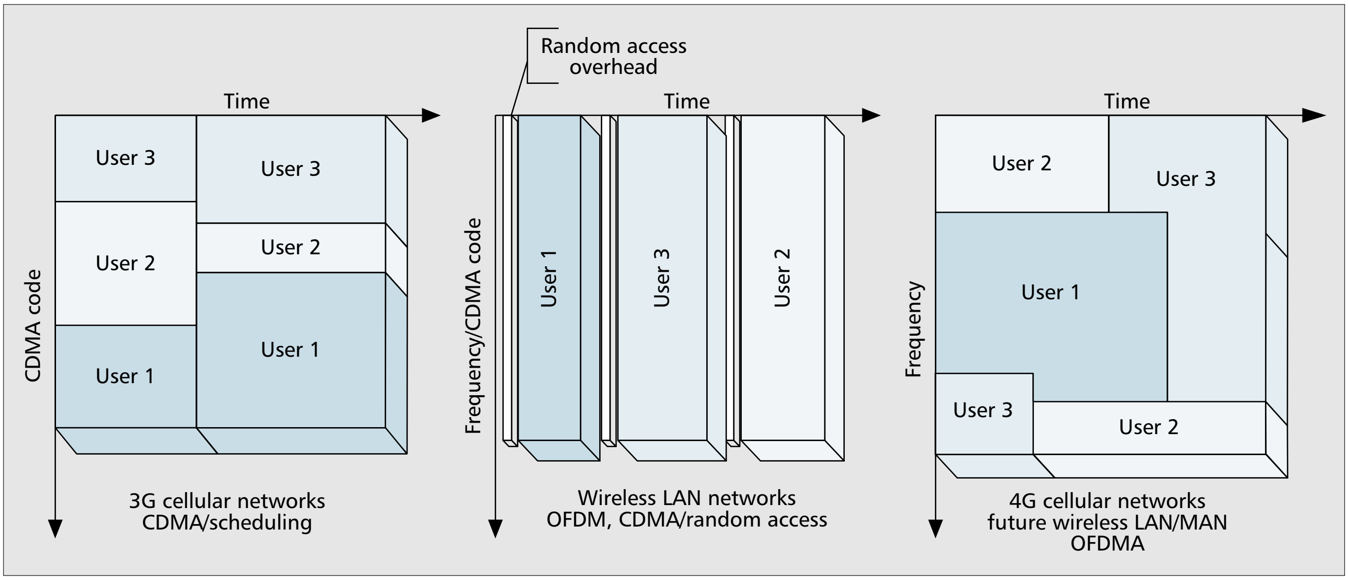
Scheduling and random access are two special types of channel-assignment schemes that enable multiple users to take turns occupying the limited radio resources over time. Scheduling is a centralized control usually applied in cellular network to determine which user can transmit at a specific time. In contrast, random access can reduce transmission delay in lightly loaded networks such as WLANs and provide an autonomous way to avoid conflict of resource usage.

In the following, we discuss the current and future broadband communication networks that can support real-time video transmission.

*3G Cellular Networks: CDMA/Scheduling* — 3G wireless communication systems employ code-division multiple access (CDMA). CDMA uses unique spreading codes to spread the baseband data before transmission. The signal occupies a much broader bandwidth than narrow-band transmission and is transmitted with power density below noise level. The receiver uses a correlator to despread the signal of interest, which is passed through a narrow bandpass filter to remove unwanted interferences. This brings many benefits, such as immunity to narrowband interference, jamming, and multiuser access.

Figure 2a shows the CDMA system with scheduler to allocate resources to users. Widely adopted in 3G networks, the scheduler allocates a different number of CDMA codes or CDMA codes with various spreading factors to users at different times according to the channel conditions, QoS types, bandwidth requirements, and buffer occupancies.

*WLAN: OFDM, CDMA/Random Access* — WLAN can provide a higher transmission rate within local areas. There are two major current standards for WLAN, namely, IEEE 802.11b and IEEE 802.11g. IEEE 802.11b uses CDMA technology and supports up to 11 Mb/s; and IEEE 802.11g uses




■ Figure 2. *Illustration of multiuser resource allocation for 3G, WLAN, 4G, and future WLAN/WMAN.*

orthogonal frequency-division multiplexing (OFDM) technology and supports up to 54 Mb/s. OFDM splits a high-rate data stream into a number of lower-rate streams and transmits them over a number of frequency subcarriers simultaneously. In addition, guard time with cyclical extension is inserted in each OFDM symbol. Thus, intersymbol and intercarrier interference are almost eliminated in OFDM systems.

Figure 2b illustrates multiple access in the current IEEE 802.11 standard. The IEEE 802.11 MAC protocol supports two access methods: the distributed coordination function (DCF) and the point coordination function (PCF). At each time slot, both functions only allow one user to occupy all radio resources. The DCF is the basic random-access mechanism using carrier-sense multiple access with collision avoidance (CSMA/CA) or ready-to-send (RTS)/clear-to-send (CTS). In contrast, the PCF is based on polling controlled by a point coordinator.

*4G Cellular Networks and Future WLAN/WMAN: OFDMA* — In current OFDM systems, all subcarriers are assigned to a single user at each moment, and multiple users are supported through time division. However, for a given subcarrier, different users experience different channel conditions and the probability for all users to have deep fades in the same subcarrier is very low. Orthogonal frequency-division multiplexing access (OFDMA) allows multiple users to transmit simultaneously on the different subcarriers, while each subcarrier is assigned to the user who is experiencing a good channel condition.

Figure 2c shows the multiuser resource-allocation strategy for OFDMA system. Users' transmission can be allocated to different time-frequency slots. By doing this, the multiuser, time, and frequency diversity can be fully explored to improve the system performance.

## Current and Future Video Coding Paradigms

Owing to the perceptual characteristics of human vision, the received video can tolerate a certain level of quality degradation. Trading lossless reconstructed quality for lossy compression can substantially reduce the required bit rate and still maintain acceptable visual quality. Most current standardized video codecs, such as H.261/3/4 and MPEG-1/2/4, adopt block-based motion-compensated prediction and block discrete cosine transform (DCT) coding with quantization to remove temporal and spatial redundancy [2]. Researchers have been also exploring 3D wavelet coding for simultaneously removing the spatiotemporal redundancies.

The potential application for video technology has evolved from precompressed files in predistributed storage, such as DVDs, to real-time encoded bit streams over wireless networks, such as video conferencing. However, there are still many remaining design challenges for real-time video compression and transmission. We summarize them as follows:

• **Perceptual Quality Control**: Unlike throughput as a major concern in data transmission systems, video systems concern video quality in terms of either subjective quality assessment or objective distortion measurement such as mean-squared error (MSE) or peak signal-to-noise ratio (PSNR). We need to control the source coding parameters in order to obtain acceptable video quality. To adapt to the variation of the scene complexity [3], the communication module needs to either dynamically adjust bandwidth or employ buffers to smoothen traffic.

• **Rate/Delay Control**: Compressed video bit streams have decoding dependency on the previous coded bit streams due to spatial and temporal prediction. Therefore, transmitting video streams in real time has a strict delay constraint that belated video data is useless for its corresponding frame and will cause error propagation for the video data that are predictively encoded using that frame as reference. We need to adjust the coding parameters to control the rate such that the bit stream can arrive at the receiver and be decoded in time.

• **Error Control**: Because of decoding dependency, video bit streams are also vulnerable to bit error, as bit error may cause the following bit stream to be decoded incorrectly. A wireless video system should take channel error into account [4]. Error-resilient tools, which increase the robustness of video bit streams, and error-concealment schemes, which utilize the received bit stream to conceal the damaged bit stream, can be integrated together to improve the end-to-end video quality [5].

• **Scalability**: A scalable video codec provides a new coding paradigm, whereby the video is encoded once and can be transmitted and decoded in many targeted rates according to the channel conditions or users' needs. Several technologies, such as fine-granularity scalability (FGS) coding and scalable video coding (SVC), have been proposed in MPEG-4 to provide spatial, temporal, and quality scalability.

## Design Principles of Multiuser Cross-Layer Resource Allocation

For video communication systems designed in layers, different layers have their own resources with practical constraints such as the feasible ranges or finite sets of discrete values that are associated with the resource parameters. For a system with cross-layer design, the allocation of system resources is constrained *vertically across layers*. For example, the bandwidth consumption for use in the application layer should not exceed the achievable capacity by the physical layer. Unlike in a single-user system, network resources are shared by multiple users in a multiuser wireless video system. Allocating these resources to one user would affect the performances of the other users due to the limited amount of resources or interference of simultaneous usage. In other words, the allocation of system resources is further constrained *horizontally among users*. Owing to the time-heterogeneity of the video source and the time-varying characteristics of the channel condition, the allocation of system resources should be performed *dynamically along time*. Moreover, real-time video transmission has additional delay constraints such as playback deadline.

A multiuser video transmission system should consider not only the video quality of each individual user but also different perspectives from the network-level point of view. In general, we can formulate the resource allocation problem so as to optimize the network objective by allocating the resources across layers and among users subject to system constraints. Two essential network objectives, *efficiency* and *fairness*, are often considered. Efficiency concerns how to attain the highest video quality summed over all users using the available system resources, and fairness concerns the video-quality deviation among users who subscribe the same QoS. There is a trade-off between efficiency and fairness.

The resource-allocation problem often has to deal with resources having both continuous and integer-valued parameters. Systems may also have nonlinear or/and nonconvex constraints and many local optima may exist in the feasible range. Thus, obtaining the optimal solution is often *NP* hard. General approaches to solve the problem are to reduce the search space by some bounds or to adaptively find the solution close to optimum. Some engineering heuristics can be employed for certain network scenarios. To allocate system resources, the resource allocators require some level of up-to-date information about available resources. Depending on the communication/computation cost and existence of central authority, systems can employ centralized or distributed algorithms. In general, resource allocators with more information can have better performance but require more communication overhead for accurate information.

Based on these design principles, in the next sections we investigate three major design methodologies for various video transmission applications over different types of wireless networks.

# Transmitting Video over 3G Cellular Networks

Transmitting video programs to users through cellular networks has become an emerging service. Video telephony and TV on demand are two such examples. Here, we consider sending multiple real-time encoded video programs to multiple mobile users over downlink CDMA systems [6].

To facilitate rate adaptation, we adopt MPEG-4 FGS. The encoder generates a nonscalable base layer at a low bit rate and an FGS enhancement layer. The decoder can decode any truncated segment of the FGS bit stream corresponding to each frame. The more bits the decoder decodes, the higher the video quality we obtain. The video codec has a *rate constraint* that the transmitted rate for each video frame should be between the base-layer rate and the maximal available FGS rate.

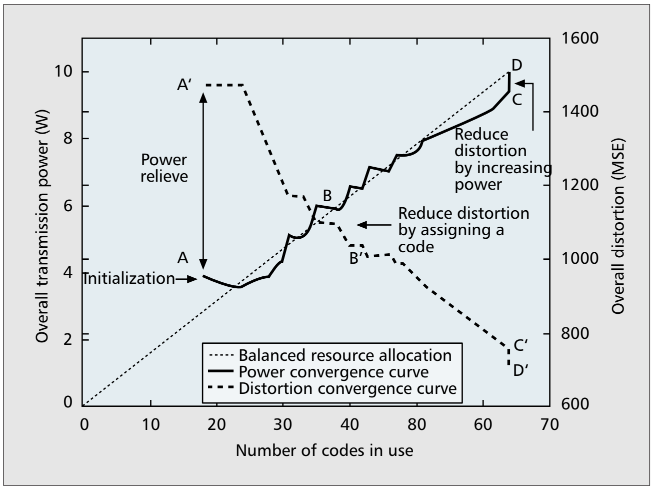
The multicode CDMA (MC-CDMA) system provides a digital bandwidth-on-demand platform by allocating multiple codes according to users' rate requests. The MC-CDMA system has the *code constraint* that a code can be assigned to at most one user. We need to determine the *code assignment*, namely, which code should be assigned to which user. The system has the *power constraint* that the overall transmission power for all codes should not be larger than the maximal transmission power, in order to limit the co-interference between cells and operate within the working range of communication circuits.

To protect bit streams from bit error during transmission, we use rate-compatible punctured convolutional code (RCPC), which provides a wide range of channel-coding rates. The goal of channel coding is to provide sufficiently low BER on the bit-stream level such that the end-to-end video quality is controllable. For MPEG-4, the degradation of video quality can be kept negligible if we enable the error-resilient features and error-concealment mechanism as well as keep the BER below a threshold set at around $10^{-6}$ [6]. To achieve the BER requirement, the received SINR should not be less than a targeted SINR that can be approximated by an exponential function of the channel-coding rate of RCPC [6].

Although a CDMA code with higher channel-coding rate can carry more source bits, the required power to meet the BER requirement is higher. The received SINR for each code is subject to interference from other codes, because of nonorthogonality among codes caused by multipath fading. Since the overall power is limited, we need to determine the *channel-coding rate assignment* of each code in order to achieve the optimal video quality subject to the power constraint and interference.

Overall, the key issue is how to jointly perform rate adaptation, code allocation, and power control so as to achieve the required perceptual qualities of received video. We consider system efficiency during each video frame-refreshing interval by determining the code assignment and channel-coding rate assignment, subject to the constraints on CDMA codes, power, and video rate. This problem is a mixed-integer programming problem, which is *NP* hard. In searching for an effectively real-time solution, we have found an important heuristic: since distortions can be reduced by using extra either power or codes, the code and power resource should be used in a balanced way so as to avoid exhausting one resource first while leaving the other resource remaining, thus leading to low system performances.

To manage the distortion, a balanced code and power-usage (BCP) algorithm is developed. An example shown in Fig. 3 illustrates how the algorithm works. We first allocate the resources for delivering the base-layer data in order to provide the baseline video quality for each user, as shown in position A. Then, we allocate resources for the FGS layer by keeping a guideline to the ratio of the current power to the number of assigned CDMA code. We assign one code at a time to a user and perform different algorithms according to the current power-to-code usage ratio. If the current ratio is larger than the ratio of the maximal power over the total number of CDMA codes (as denoted by the dotted line), the system consumes higher than average power per code. Position A is one of the examples. In this case, a new code is




■ Figure 3. *Power (solid line) and distortion (dashed line) convergence track vs. the number of assigned codes. The maximal power is 10 W and the total number of codes is 64. Positions A, B, C, and D are examples indicating the overall power when different numbers of codes are assigned, and positions A', B', C', and D' are the corresponding overall distortion.*

assigned to a user by keeping the user's source coding rate unchanged but redistributing the source rates among user's already assigned codes plus the new code. Subsequently, the channel-coding rates for those codes, the required SINR, and the overall power are all reduced. If the current power-to-code ratio is smaller than the ratio of the maximal transmission power over the total number of CDMA codes, the system consumes moderate power per code. One such example is position B. In this case, we will assign a new code with maximal channel-coding rate to carry video source bits such that the overall distortion will be reduced. By doing so, the total power consumption will increase. After all codes are assigned, we perform a round of refinement to further reduce the distortion by using the remaining power quota. Position C to position D is an example of quality refinement. At the end, all available power and code resources are fully utilized.

We compare BCP algorithm to a greedy algorithm. For each iteration, the greedy algorithm tries to assign a candidate code with a maximal channel-coding rate to every user, calculates the distortion reduction, and assigns a new code to the user with the largest value. We concatenate 15 classic QCIF ($176 \times 144$) video sequences to form a basic testing sequence of 2775 frames [6]. The content program for each user is 100 frames and starts from a randomly selected frame of the testing sequence. The location for each user is uniformly distributed within the cell with radius from 20 to 1000 m. The video-refreshing rate is 15 frames per second (fps). Figure 4 shows the comparison result for different number of users versus the total distortion, $D_{sum}$. The simulation results demonstrate that the BCP algorithm outperforms the greedy algorithm by 14 to 26 percent. The main reason for this gain is that the greedy algorithm ignores the balance between power and code usages and thus depletes one resource while wasting other resources.

## Video Conferencing over WLANs

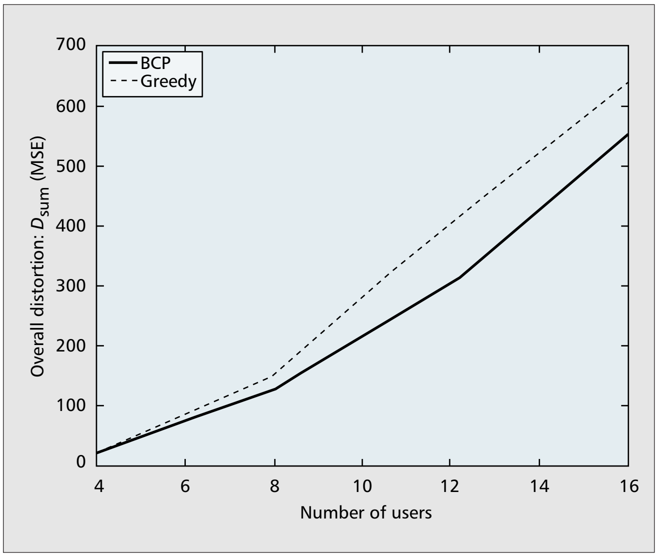
With the wide deployment of WLANs, transmitting real-time video such as video conferencing to mobile users has become an attractive service [7, 8]. In this section, we present a real-time interactive video conferencing framework to support multiple video conferencing pairs [9]. For each conversation pair, there are two video streams exchanged and each video stream is transmitted through an uplink and a downlink. Our framework explores the diversity of video content and the heterogeneity of uplink and downlink channel conditions experienced by different users over the resource-limited wireless networks.

The IEEE 802.11a PHY layer provides eight operation modes using different modulation schemes and convolutional coding rates. Different PHY modes have different BER performance and data rates, ranging from 6 to 54 Mb/s. In this system, we need to determine the *PHY mode assignment*, namely, select which PHY mode of uplink and downlink for each user can achieve the optimal video quality.

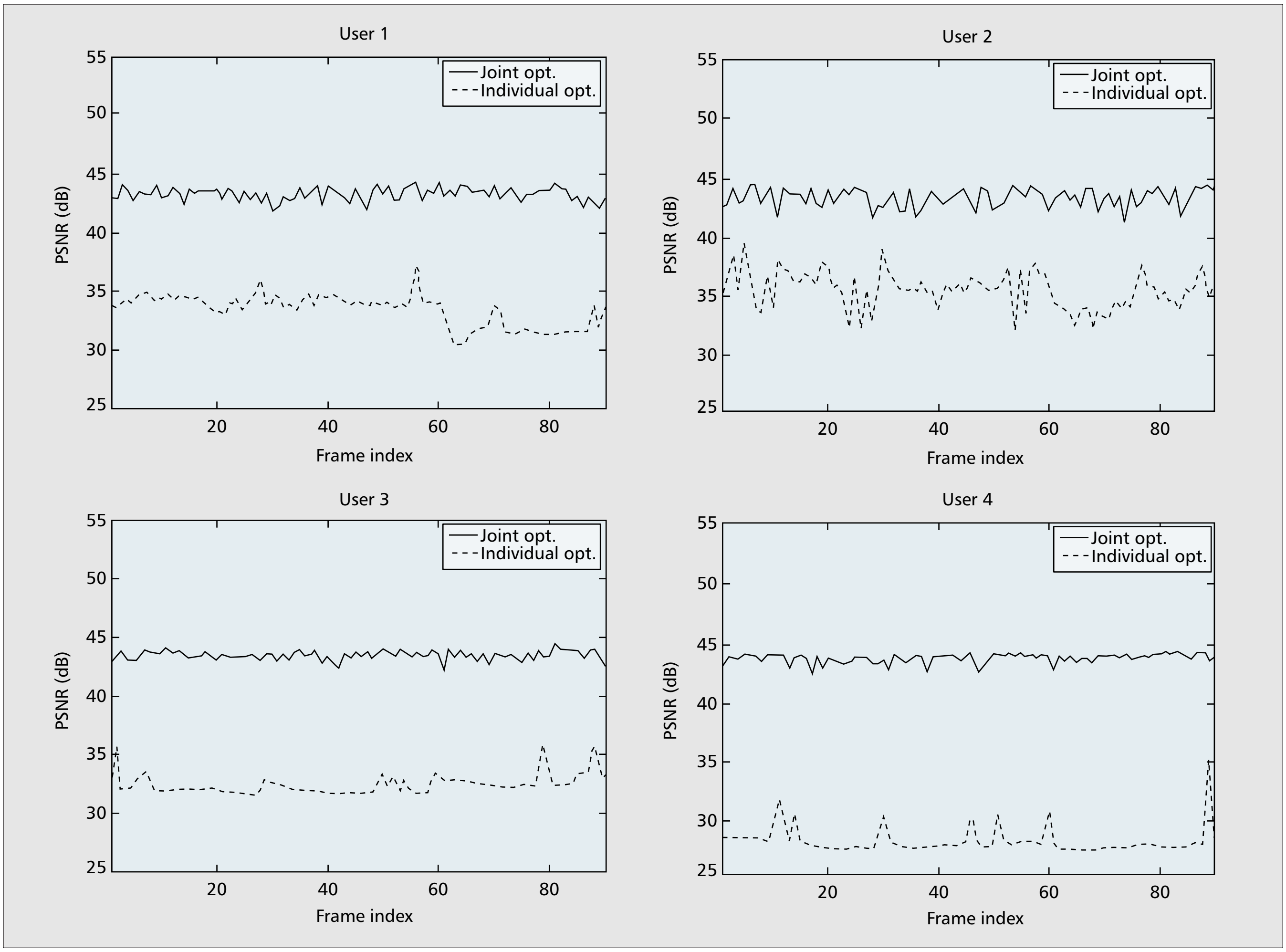
The IEEE 802.11 MAC protocol only allows one user occupy the entire bandwidth at each time. Each individual user's transmission time can be controlled by either the PDF or the enhanced DCF. For the real-time video transmission scenario, all users should transmit their video frames within each frame-sampling period. Consequently, the system has the *transmission time constraint* that the overall transmission time of all users is bounded by this period. In this work, we study how to optimize the amount of transmission time allocated to each user under this transmission time constraint.

The transmission errors can be detected by checking the cyclic redundancy check bits of a packet. If the server finds the packet is corrupted by errors, this packet will not be forwarded to its destination, which leads to packet loss. With the transmission index, we can identify which packet is lost in the receiver. Thus, we can model this channel as an erasure channel. Applying application-layer FEC across packets, such as using Reed-Solomon (RS) codes, has been shown to be an effective solution to alleviate the problem caused by packet loss. For the source coding part, we adopt MPEG-4 FGS to facilitate rate adaptation. To jointly determine the source coding and channel protection, we need to determine the *packet assignment*, namely, the number of source packets and parity check packets for each video frame.

In this system, we consider fairness among users by choosing the PHY mode assignment and packet assignment subject to the transmission time constraint. Searching the optimal setting is *NP* hard. A two-stage algorithm is developed to obtain a near-optimal solution. First, for each user, we map the required system resources and the corresponding expected distortion into a function relating to the transmission time and




■ Figure 4. *Performance comparison of BCP and greedy algorithms.*




■ Figure 5. *Frame-by-frame PSNR results.*

the expected distortion (we call this function the T-D function). By doing this, the traditional R-D function in single-user transmission evolves to a resource-distortion function in the multiuser scenario. Second, a round of a bisection search on all users' T-D functions is performed to find the solutions.

We compare our joint optimization scheme with a traditional scheme that allocates equal transmission-time budgets for uplink and downlink, and sequentially optimizes these two links. We consider a four-user system, where users 1 to 4 are located 91, 67, 71, and 20 m away from the server, and receive a 90-frame QCIF sequence of *Akiyo*, *carphone*, *Claire*, and *foreman*, respectively. Figure 5 shows the frame-by-frame PSNR results. We can see that the new scheme has higher average PSNR, fairer video quality among all users, and less quality fluctuation along each received video sequence than the traditional scheme.

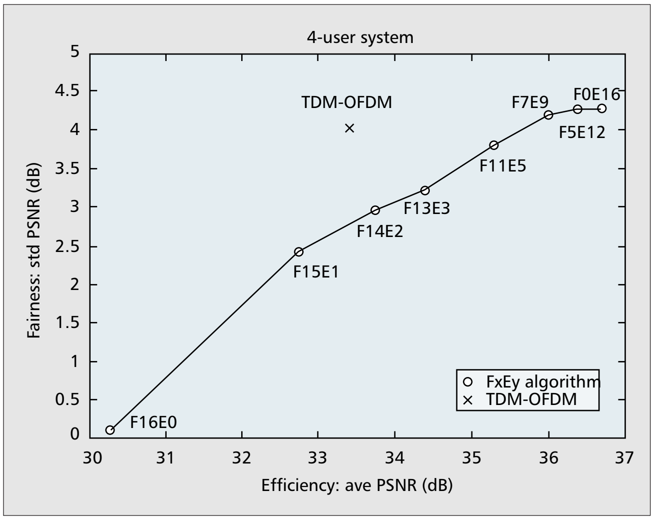
## Next-Generation Video over OFDMA Networks

In this section, we discuss next-generation wireless video transmission both in terms of wireless communication networks and the video codecs. We consider the downlink scenario of streaming multiple embedded video bit streams over a single-cell OFDMA system [10].

We use 3D embedded wavelet video as an example of a scalable video codec. The encoder collects a group of frames (GOF) as a coding unit, performs 3D wavelet transform to obtain spatiotemporal subbands ("subbands" for short), and encodes each subband into several coding passes. The more consecutive coding passes of each subband a receiver receives, the higher the decoded video quality we have. The coding passes among all subbands can be further grouped into several quality layers such that the received video quality can be refined progressively. We need to determine *coding-pass assignment*, namely, which coding pass should be assigned to which quality layer for each user.

Assume the channel condition is stationary within a given period of time, referred to as the transmission interval. We can divide the maximal allowed time duration to transmit an encoded GOF bit stream (for meeting the delay constraint) into $L$ transmission intervals of equal length. For each video, we dynamically construct and transmit a quality layer in each transmission interval. The total allocated rate for each quality layer has the *rate constraint* that it cannot exceed the available bandwidth.

A single-cell OFDMA system has the *subcarrier constraint* that each subcarrier can be assigned to at most one user. For a given subcarrier, users may experience different channel conditions and require different power to achieve the same signal-to-noise ratio (SNR). To overcome channel errors, we maintain a BER after FEC decoding below $10^{-6}$ such that the end-to-end quality is controllable. The supported rate of a subcarrier is a function of the required SNR maintaining such a BER through adaptive modulation and coding. A subcarrier with a higher power level can have a higher transmission rate. As the overall power for all subcarriers is limited, this system




■ Figure 6. *Efficiency and fairness of the FxEy algorithm family and TDM-OFDM algorithm.*

has a *power constraint*. We need to jointly determine which subcarrier should be assigned to which user (*subcarrier assignment*) and the subcarrier rate in each subcarrier (*rate assignment*) subject to power constraint.

We consider both the fairness and efficiency problems by determining the assignments on coding pass, rate, and subcarrier, subject to constraints on subcarrier, rate, and power. Both problems are mixed-integer optimization problems, which are again *NP* hard. For real-time purposes, we develop a fairness algorithm and an efficiency algorithm, both of which have the following two steps. We first obtain an operational R-D function of all unsent coding passes for the current GOF of each user. The R-D function specifies the required data rates for reducing a certain amount of distortion. Then, we search the R-D functions and find the highest required rate that can be supported by the OFDMA system.

There is an inherent trade-off between fairness and efficiency. To achieve this trade-off, for each GOF, we apply the fairness algorithm in the first $x$ transmission intervals to ensure the baseline fairness, and then apply the efficiency algorithm in the remaining $y = L - x$ transmission intervals to improve the overall video quality. We denote this class of strategies as $F_xE_y$ algorithms.

We examine the performance of the $F_xE_y$ strategies as follows. We concatenate 15 classic CIF ($352 \times 288$) sequences to form one testing sequence of 4064 frames [10]. The video for each user is 160 frames and starts from a randomly selected frame of the testing sequence. The GOF size is 16 frames and the transmission interval is 33.33 ms. The mobile is uniformly distributed within the cell with radius from 10 to 50 m. Figure 6 shows the fairness (measured in terms of PSNR deviation) and efficiency (measured in terms of average of PSNR) for the $F_xE_y$ algorithms. Within the $F_xE_y$ algorithm family, the $F_{16}E_0$ algorithm achieves the lowest PSNR deviation but has the lowest average PSNR; and the $F_0E_{16}$ algorithm achieves the opposite performance. We can have different trade-offs of these two criteria by adjusting parameter x. Next, we compare the $F_xE_y$ algorithms with a conventional TDM-OFDM algorithm, which assigns all subcarriers in one transmission interval to only one user whose current distortion is the largest so as to maintain fairness in the min-max sense. As shown in Fig. 6, for achieving the same average PSNR, our algorithm is about 1 dB lower in PSNR deviation than the TDM-OFDM algorithm. When achieving the same PSNR deviation, our algorithm can have higher average PSNR than the TDM-OFDM algorithm. This is because our scheme exploits additional diversity in frequency and multiuser.

## Conclusions

We have discussed multiuser video transmission over wireless networks from the resource allocation point of view. In comparison to the traditional schemes, we have considered two design issues: cross-layer optimization and multiuser diversity. A general resource-allocation framework was presented to control the quality of multiple video transmissions that share limited resources over wireless networks. We reviewed several schemes for transmitting video programs over 3G, 4G, and WLAN/WMAN systems and demonstrated that multiuser cross-layer design can provide better video quality than the traditional schemes.

## References


[1] A. K. Katsaggelos *et al.*, "Advances in Efficient Resource Allocation for Packet-Based Real-Time Video Transmission," *Proc. IEEE*, vol. 93, no. 1, Jan. 2005, pp. 135–47.

[2] T. Sikora, "Trends and Perspectives in Image and Video Coding," *Proc. IEEE*, vol. 93, no. 1, Jan. 2005, pp. 6–17.

[3] M.-T. Sun and A. R. Reibman, *Compressed Video Over Networks*, Marcel Dekker, 2001.

[4] K. Stuhlmuller *et al.*, "Analysis of Video Transmission over Lossy Channels," *IEEE JSAC*, vol. 18, no. 6, June 2000, pp. 1012–32.

[5] Y. Wang, J. Ostermann, and Y.-Q. Zhang, *Video Processing and Communications*, Prentice Hall, 2001.

[6] G.-M. Su *et al.*, "Distortion Management of Real-Time MPEG-4 FGS Video over Downlink Multicode CDMA Networks," *IEEE Int'l. Conf. Commun.*, vol. 5, June 2004, pp. 3071–75.

[7] M. van der Schaar *et al.*, "Adaptive Cross-Layer Protection Strategies for Robust Scalable Video Transmission over 802.11 WLANs," *IEEE JSAC*, vol. 21, no. 10, Dec. 2003, pp. 1752–63.

[8] A. Majumda *et al.*, "Multicast and Unicast Real-Time Video Streaming over Wireless LANs," *IEEE Trans. Circuits Syst. Video Technol.*, vol. 12, no. 6, June 2002, pp. 524–34.

[9] G.-M. Su *et al.*, "Joint Uplink and Downlink Optimization for Video Conferencing over Wireless LAN," *IEEE Int'l. Conf. Acoustics, Speech, and Sig. Proc.*, vol. 2, Mar. 2000, pp. 1101–04.

[10] G.-M. Su *et al.*, "Dynamic Distortion Control for 3-D Embedded Wavelet Video over Multiuser OFDM Networks," *IEEE Global Telecommun. Conf.*, vol. 2, Nov. 2004, pp. 650–54.


## Biographies


GUAN-MING SU (gmsu@eng.umd.edu) received his B.S.E. degree from National Taiwan University, Taiwan, in 1996 and his M.S. degree from the University of Maryland, College Park, in 2001, both in electrical engineering. He is currently working toward his Ph.D. degree at the University of Maryland, College Park. He was with the R&D Department, Qualcomm, Inc., San Diego, California during the summer of 2005. His research interests are multimedia communications and multimedia signal processing.

ZHU HAN (hanzhu@eng.umd.edu) received his B.S. in electronic engineering from Tsinghua University, 1997, and M.S. and Ph.D. degrees in ECE from the University of Maryland, College Park, 1997 and 2003. From 1997 to 2000, he was a GRA at University of Maryland. From 2000 to 2002, he was an engineer in the R&D Group of ACTERNA. He is currently a research associate at the University of Maryland. His research interests include wireless networking and resource allocation, game theory, and wireless multimedia.

MIN WU (minwu@eng.umd.edu) received B.E. and B.A. degrees from Tsinghua University, China, in 1996, and her Ph.D. degree in electrical engineering from Princeton University in 2001. She is an assistant professor with the Electrical and Computer Engineering Department at the University of Maryland, College Park. Her research interests include information security and forensics, multimedia signal processing, and multimedia communications. She is an ONR Young Investigator and received an NSF CAREER award, a TR100 Young Innovator award, and two best paper awards.

K. J. RAY LIU (kjrliu@eng.umd.edu) [F'03] is a professor of the Electrical and Computer Engineering Department, University of Maryland, College Park, where he received the Poole and Kent Senior Faculty Teaching Award and Invention of the Year Award. He has received best paper awards from IEEE and EURASIP. He is Vice President–Publications and was a Distinguished Lecturer of IEEE Signal Processing Society. He was Editor-in-Chief of *IEEE Signal Processing* and *EURASIP Journal on Applied Signal Processing*.