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Multiuser Distortion Management of Layered Video over Resource Limited Downlink Multicode-CDMA

Zhu Han, Guan-Ming Su, Andres Kwasinski, Min Wu, and K. J. Ray Liu

***Abstract*— Transmitting multiple real-time encoded videos to multiple users over wireless cellular networks is a key driving force for developing broadband technology. We propose a new framework to transmit multiple users' video programs encoded by MPEG-4 FGS codec over downlink multicode CDMA networks in real time. The proposed framework jointly manages the rate adaptation of source and channel coding, CDMA code allocation, and power control. Subject to the limited system resources, such as the number of pseudo-random codes and the maximal power for CDMA transmission, we develop an adaptive scheme of distortion management to ensure baseline video quality for each user and further reduce the overall distortion received by all users. To efficiently utilize system resources, the proposed scheme maintains a balanced ratio between the power and code usages. We also investigate three special scenarios where demand, power, or code is limited, respectively. Compared with existing methods in the literature, the proposed algorithm can reduce the overall system's distortion by 14% to 26%. In the demand-limited case and the code-limited but power-unlimited case, the proposed scheme achieves the optimal solutions. In the power-limited but code-unlimited case, the proposed scheme has a performance very close to a performance upper bound.**

***Index Terms*— Resource management, multimedia communication, code division multiaccess, land mobile radio cellular systems, adaptive coding.**

I. Introduction

OVER the past few decades, wireless communications and networking have experienced an unprecedented growth. Wireless networking has become ubiquitous owing to the great demand of pervasive mobile applications such as video transmission [1], [2]. Code Division Multiple Access (CDMA) is a promising technology used in the current wireless networks and widely adopted for 3G wireless networks. The challenges for transmitting compressed videos over CDMA in real time lie in several aspects. First, CDMA networks are interference limited, and there are limited radio resources such as the transmission power and the total number of pseudo-random codes. In order to accommodate a large number of users with acceptable received quality, one challenge for system design is how to optimally allocate these radio resources. Furthermore, the bit rate of compressed video can be highly bursty due to the difference in video contents and intra/inter coding modes. To achieve the desirable video quality, we need to adjust the system parameters, such as source coding and radio resource allocations, to each video stream dynamically.

Transmitting pre-compressed videos with constant bit rate (CBR) [3] can be considered as one of the simplest video over CDMA systems due to the simplicity of radio resource allocation. On the other hand, a video encoded in variable bit rate (VBR) bitstream gives better perceptual quality for end users than in CBR bitstream due to the variation of the scene complexity [4]. However, VBR traffic exhibits a highly bursty rate, which requires a communication system to support variable transmission rate. Furthermore, current CDMA systems such as IS-95 cannot accommodate high data rate needed for transmitting high quality video. Multicode CDMA (MC-CDMA) system [5], [6] provides a digital bandwidth-on-demand platform by allocating multiple codes on demand and hence providing increased capacity to users. Several schemes have been proposed for transmitting pre-compressed VBR videos over MC-CDMA system [7]–[9], subject to the limited number of pseudo-random codes.

In designing a video transmission system, we also concern the adjustability of the video codec, namely, how easily a video system can change video encoding rate to achieve different desired visual qualities in real time, subject to the constraints of system resources. For a system with pre-stored videos, we can pre-calculate the received quality of a pre-stored video sequence using different source bit rate and bit-error probability through simulation and decide the actual bit rate to transmit [10]. For a real-time compressed video, the values of the video codec's parameters should be selected on the fly to control the desired rate or quality [11],

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Z. Han is with the Department of Electrical and Computer Engineering, Boise State University, Boise, ID (e-mail: zhuhan@boisestate.edu).

G.-M. Su is with ESS Technology, Fremont, CA 94538 USA (email: guanming.su@esstech.com).

A. Kwasinski, M. Wu, and K. J. R. Liu are with the Department of Electrical and Computer Engineering and the Institute of Systems Research, University of Maryland, College Park, MD (email: {ak, minwu, kjrliu}@eng.umd.edu).

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[12]. However, the computational complexity of searching the optimal solutions for the traditional non-scalable video is prohibitive. A highly scalable video codec, such as MPEG-4 Fine Granularity Scalability (FGS) coding [13] and Fine Granular Scalability Temporal (FGST) coding [14], is desirable since it provides flexibility and convenience in reaching the desired visual quality and/or the desired bit rate. We will use MPEG-4 FGS in our work. Furthermore, a multiuser video system should support transmitting heterogeneous video sequences to different user simultaneously. It has been shown that jointly allocating coding rate to multiple video sources can leverage the difference in video content complexities to improve the resource utilization and achieve more desired quality [15]–[17].

The resource allocation for maximizing overall throughput for multiple users has been widely studied [18]–[20]. However, a wireless real-time video system supporting maximal throughput cannot provide the required video quality for all users due to the delay requirement and different content complexity. Jointly considering power control with source/channel coding has been shown as an effective way to improve the received multimedia quality [21], [22]. Algorithms for a single-user system have been developed to either minimize the overall power consumption of video source encoding, channel coding, and transmission subject to video quality constraint [23]–[25], or to minimize the expected received video distortion subject to transmission power constraint [26], [27]. For video transmission over CDMA network, it is important to perform power control among all users as CDMA system is interference-limited [28].

Few video transmission works in literature consider the case that multiple users share the limited network resources, where the allocation of resources to one user would affect the performances of the other users. Moreover, mixed integer optimization is required in practice, since most parameters such as coding rate have finite sets of discrete values, while the continuous relaxations of parameters are often assumed in the cross-layer literature. So this motivates us to propose a multiuser cross-layer framework to bring important insights on designing a resource-limited multiuser video transmission system. Specifically, for the source coder subsystem, we select MPEG-4 FGS video because of its fine-granular scalability in the enhancement layer. For the channel coder subsystem, we adopt the rate compatible punctured convolutional (RCPC) code [29] because of its wide range of channel coding rates and simplicity in design and implementation. For the wireless communication subsystem, we use the MC-CDMA system [5], [6] as it can provide bandwidth according to the users' demand.

The key issue is how to jointly perform the rate adaptation, code allocation, and power control to minimize the overall end-to-end distortion received by all users, subject to the limited available radio resources such as the number of codes and transmission power. We develop a distortion management algorithm to allocate resources to users by balancing the code and power usages. To study the performance of the proposed algorithm, we analyze three special cases, namely, the demand-limited, code-limited, and power-limited cases. For the demand-limited case, a close-form solution is provided. For the code-limited case, the original problem is reformulated and the optimal solution is obtained. For the power-limited case, a performance upper bound is derived. Simulations show that the proposed algorithm reduces the distortion by $14 \sim 26\%$, when compared with the modified greedy algorithm [30]. The proposed scheme achieves the optimal solutions when the system operates in the demand-limited and code-limited cases, and performs close to the performance upper bound when the system operates in the power-limited case.

This paper is organized as follows. The designed system and the available system resources are presented in Section II. In Section III, we formulate this system as an optimization problem and develop a resource allocation algorithm. In Section IV, three special cases are studied. Simulations are presented in Section V and conclusions are drawn in Section VI.

II. System Description

In this section, we first present an overview of our proposed system to transmit multiple MPEG-4 FGS video streams over downlink MC-CDMA. We then describe the video source coder subsystem and MC-CDMA transmission subsystem in details. For each subsystem, we will discuss the available resources, study how to control parameters to achieve the desired result, and analyze the corresponding constraints in the practical implementation.

A. System Overview

Figure 1 shows the block diagram of the proposed wireless video server located at the base station, which transmits multiple real-time encoded video programs to multiple mobile users. There are three major subsystems in the proposed system: video source coding subsystem, MC-CDMA subsystem, and resource allocation subsystem for managing distortion.

The resource allocation subsystem first collects the necessary information from the video source coders and MC-CDMA subsystem. In the video source coding subsystem, each video program is encoded by a FGS encoder in real time. These FGS encoders compress the incoming video frames and send the corresponding R-D information to the resource allocation subsystem. The downlink channel information is obtained via feedback from mobiles. Considering that the network provides the heterogenous services, the system allocates some available radio resources such as transmission power and CDMA codes for video transmission. In this paper, we assume the amount of these resources is known.

After gathering the source and channel information, the resource allocation subsystem executes optimization algorithms and allocates system resources to achieve the system optimization objectives. Interfacing with the video source coder subsystem, the resource allocation subsystem adjusts video encoders' source rates; and interfacing with the MC-CDMA system, the resource allocation subsystem assigns a variable number of codes to each user according to his/her resource needs and channel conditions. For example, an I frame requires more CDMA codes to be assigned than a P frame. In addition, the resource allocation subsystem determines channel coding adaptation and power control for each code to protect the

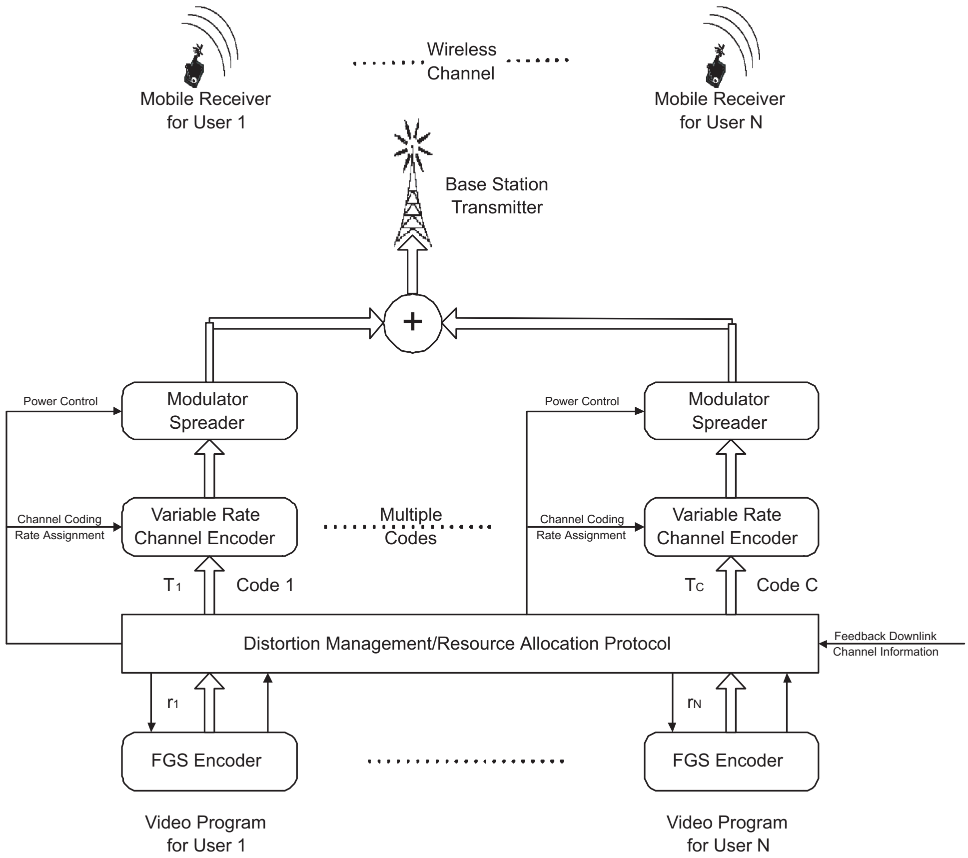

Fig. 1. Block diagram for the proposed protocol.

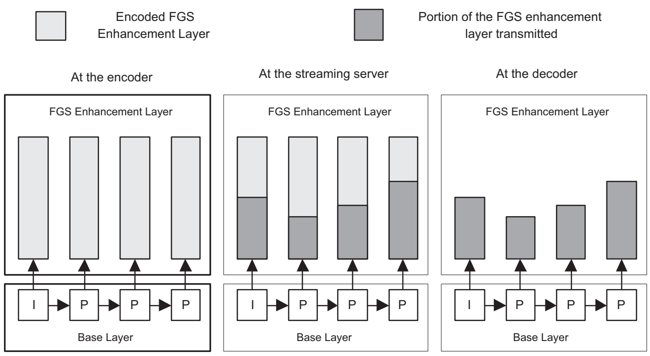

Fig. 2. Streaming video using MPEG-4 FGS.

transmitted data. In allocating resources, our goal is to maintain good video qualities, even when transmitting through a noisy channel with interference. As channel-induced bit errors may affect video qualities in an unpredictable way, to avoid the uncertainty and maintain controllable video qualities, we use adaptive channel coding and power control to achieve a sufficiently small bit error rate (BER).

B. Video Source Coder Subsystem

As mentioned in the introduction, the MPEG-4 Fine Granularity Scalability (FGS) coding [13] and Fine Granular Scalability Temporal (FGST) coding [14] are scalable video techniques for delivering streaming video. FGS coding enables a video sequence to be encoded once, and transmitted/decoded at different rates according to the available bandwidth. The encoder generates a base layer at a low bit rate using a large quantization step and computes the residues between the original frame and the base layer. The bit planes of Discrete Cosine Transform (DCT) coefficients of these residues are encoded sequentially to form the FGS enhancement layer. After completely decoding the base layer, the decoder can decode any truncated segment of the FGS bitstream corresponding to each frame. The more bits the decoder receives and decodes, the higher the video quality is. Figure 2 illustrates a streaming video system using a FGS codec. The encoder encodes all bit planes for each video frame and lets the video server determine how many bits to send for each frame according to the channel condition.

Existing video schemes using a single-layer video codec often explicitly employ R-D models and an exponential or a polynomial R-D model is frequently used [16], [31]. The MPEG-4 FGS codec is a two-layer scheme and its enhancement layer is encoded bit plane by bit plane. For a given bit plane in a frame, if the video is spatially stationary so that the length of the entropy encoded FGS symbols in all blocks is similar to each other, the decoded bit rate and the corresponding amount of reduced distortion will have an approximately linear relationship over the bit rate range of this bit plane. Previous studies in [16], [32] show that a piecewise linear function is a good approximation to the R-D curve of FGS video at the frame level. This piecewise linear function model can be summarized as:

$$D_{n,j}(r_{n,j}) = M_{n,j}^k(r_{n,j}-R_{n,j}^k)+E_{n,j}^k,\ k=0,\ldots,p-1, \quad (1)$$

$$\text{with } M_{n,j}^k = \frac{E_{n,j}^{k+1}-E_{n,j}^k}{R_{n,j}^{k+1}-R_{n,j}^k}, R_{n,j}^k \le r_{n,j} \le R_{n,j}^{k+1},$$

where p is the total number of bit planes, and $E_{n,j}^k$, $R_{n,j}^k$, and $r_{n,j}$ denote the distortion measured in mean square error (MSE) after completely decoding the first k DCT bit planes, the corresponding bit rate, and the overall decoded bit rate for the j^{th} user's distortion of the n^{th} frame, respectively. Note that $E_{n,j}^0$ and $R_{n,j}^0$ represent the distortion and source rate of the base layer, respectively. Because DCT is a unitary transform, all $(R_{n,j}^k, E_{n,j}^k)$ R-D pairs including the base layers can be obtained during the encoding process without decoding the compressed bitstreams. Since we allocate resources frame by frame, we omit n from the notation for simplicity.

C. MC-CDMA Transmission Subsystem

Consider a single-cell MC-CDMA system with N users and a total of C codes for video transmission. We assume the downlink system is synchronous and each user is assigned a set of unique fixed-length pseudo-random codes. Because of the multipath effect [33], the orthogonality among codes may not be guaranteed. Consequently, each mobile user is subject to interference from other users in the cell. If the i^{th} code is assigned to user j, the received signal-to-interference-and-noise ratio (SINR) for this code is:

$$\Gamma_i^j = \frac{W}{R}\frac{P_iG_j}{G_j\sum_{k=1,k\neq i}^{C}\alpha_{ki}P_k+\sigma^2}, \quad (2)$$

where W is the total bandwidth and is fixed, R is the transmission rate, P_i is the transmission power from the base station for code i, α_{ki} is orthogonality factor between the k^{th} and i^{th} codes and can be estimated statistically [33], G_j is the j^{th} user's path loss which is assumed to be stable within each video frame and can vary from frame to frame, and σ^2 is the thermal noise level and assumed to be the same at all mobile receivers. The ratio W/R is the processing gain. In this paper, we use BPSK modulation for simplicity.

For a fixed transmission rate per code in a CDMA system, a reduction of source bits being carried allows for inserting more channel error protection. In order to jointly adjust rates in both source and channel coders according to the needs for distortion controls and channel error protections, we consider channel coders with adjustable rates. In this paper, an RCPC [29] is applied for channel coding, because of its wide range of channel coding rates and simplicity in implementation. A family of RCPC codes is described by the mother code of rate $\frac{1}{M}$. The output of the coder is punctured periodically following a puncture table. The puncturing period Q determines the range of channel coding rates $r=\frac{Q}{Q+l},\ l=1,\ldots,(M-1)Q$, which are between $\frac{1}{M}$ and $\frac{Q}{Q+1}$ with decreasing channel error protection ability. The rate compatibility ensures that the codes with the lower rates contain the codes with the higher rates. Consequently, the progressive transmission is possible by simply transmitting the additional information, when more channel error protection becomes necessary. Moreover, only one Viterbi receiver is needed for the RCPC codes with different rates, which reduces the system complexity.

Our goal is to maintain good end-to-end subjective video quality. Since the channel-induced distortion is typically more annoying than the distortion introduced by source encoding, the system keeps the channel-induced distortion at a small proportion of the end-to-end distortion such that the video quality is controllable by the source coding subsystem. This can be achieved by enabling the error resilient and error concealment features in the source coding subsystem and maintaining the BER after channel decoding to be below a threshold in the communication subsystem. To reach the BER requirement, the actually received SINR should be no less than some threshold. We shall refer to this threshold as the targeted SINR. Through simulations using RCPC codes [29], we have found that to achieve the BER threshold, the targeted SINR can be very well approximated as a function of channel coding rate by [34]

$$\gamma_i = 2^{AT_i+B}, \quad (3)$$

where γ_i is the required targeted SINR, T_i is the channel coding rate with discrete value in the range of $[T_{min}, T_{max}]$, and A and B are the parameters related to the channel coding that can be experimentally determined. Their values depend on the BER requirements, fading conditions, and channel code family being used.

Given a set of channel coding rates, we can compute the overall required transmission power for all users, P_{sum}, by bringing in (2) and (3) together:

$$P_{sum} = \sum_{i=1}^{C} P_i = \mathbf{1}^T[\mathbf{I}-\mathbf{F}]^{-1}\mathbf{u}, \quad (4)$$

where $\mathbf{1}=[1\ldots1]^T$, $\mathbf{u}=[u_1,\ldots u_C]^T$ with $u_i=\sigma^2Y_i/G_i$, and $[\mathbf{F}]_{ii'}=0$, if $i'=i$; $[\mathbf{F}]_{ii'}=\alpha_{i'i}Y_i$, if $i'\neq i$. Here Y_i is defined as follows, depending on which user the code is assigned to:

$$Y_i = \begin{cases} 0, & \text{if code is not assigned;} \\ \frac{2^{AT_i+B}R}{W}, & \text{if code } i \text{ is assigned to user } j. \end{cases} \quad (5)$$

Since the processing gain W/R is large, Y_i is small, thus P_{sum} can be approximated as:

$$P_{sum} \approx \mathbf{1}^T[\mathbf{I}+\mathbf{F}]\mathbf{u} = \sum_{i=1}^{C}\frac{\sigma^2Y_i}{G_i} + \sum_{i=1}^{C}\sum_{k\neq i}^{C}\frac{\sigma^2\alpha_{ki}Y_iY_k}{G_k}. \quad (6)$$

Note that the overall transmission power P_{sum} is bounded by the available transmission power, and it is necessary to perform power control among all users and the available CDMA codes.

III. DISTORTION MANAGEMENT ALGORITHM

In this section, we formulate the distortion management problem. We then present the proposed two-stage algorithm to transmit the base layer first and then the FGS enhancement layer to fully utilize the limited resources.

A. Problem Formulation

In this MC-CDMA system, we denote $a_{ij} \in \{0, 1\}$ as an indicator to specify whether the i^{th} code is assigned to user j. The total available power for video transmission is P_{max}, and each user's throughput, r_j, should be larger than the base layer rate R_j^0 to guarantee the baseline quality and should be smaller than the maximum source rate R_j^P. The transmission rate for each CDMA code is R. Each code will carry information of rate RT_i, where T_i is the channel coding rate for the i^{th} code. We formulate this problem as to minimize the overall system distortion under the code, power, and rate constraints:

$$\min_{T_i, a_{ij}} \sum_{j=1}^{N} D_j(r_j) \tag{7}$$

$$\text{s.t.} \begin{cases} \text{code constraint: } \sum_{j=1}^{N} a_{ij} \le 1, a_{ij} \in \{0,1\}, \forall i; \\ \text{power constraint: } P_{sum} = \sum_{i=1}^{C} P_i \le P_{max}; \\ \text{rate constraint: } R_j^0 \le r_j = R\sum_{i=1}^{C} a_{ij} T_i \le R_j^P, \forall j. \end{cases}$$

Notice that the BER requirement and the corresponding minimal SINR are already implicitly ensured by (3). The problem in (7) is a mixed integer programming problem with nonlinear and non-convex constraints in rate and power. Although an exhaustive search algorithm can obtain the optimal solution, the computational complexity over the large search space is prohibitive. Further, as we will discuss later, even a simplified version of the above problem is still NP hard. A fast suboptimal algorithm is thus preferable for real-time applications.

In the next two subsections, we will develop algorithms to efficiently allocate the limited power and codes to reduce the overall distortion. There are two main stages in our proposed algorithm. At the first stage, we allocate the resources for delivering the base layer data to provide the baseline video quality for each user. Some FGS enhancement layer data is then delivered to reduce the overall distortion by allocating the remaining resources and making necessary adjustment on the resource allocation. Our resource allocation strategy aims at using the code and power resources in a balanced way so as to avoid exhausting one resource first while having the other resource left, which leads to undesired local optima.

B. Resource Allocation for Base Layer

The upper part of Figure 3 shows the initialization procedure, where codes, channel coding rates, and power are assigned to each user so that all users can receive the base layer rate of R_j^0 under the power constraint. First, we set the channel coding rate to the highest level T_{max} as offered by the adopted RCPC code and assign a total of $C_j = \lceil \frac{R_j^0}{RT_{max}} \rceil$ codes to each user, such that all the base layers can be transmitted. If there is no enough codes for this assignment, an outage will be reported, indicating that there are too many users in the system and there are no resources even for accommodating the base layers only. If there are enough codes for the base layer, the proposed system then determines whether the power constraint is violated. If not, the initialization is done and the system starts running the main algorithm for allocating resources for the FGS layer. Otherwise, we will relieve P_{sum} by assigning one code at a time to the user who can reduce

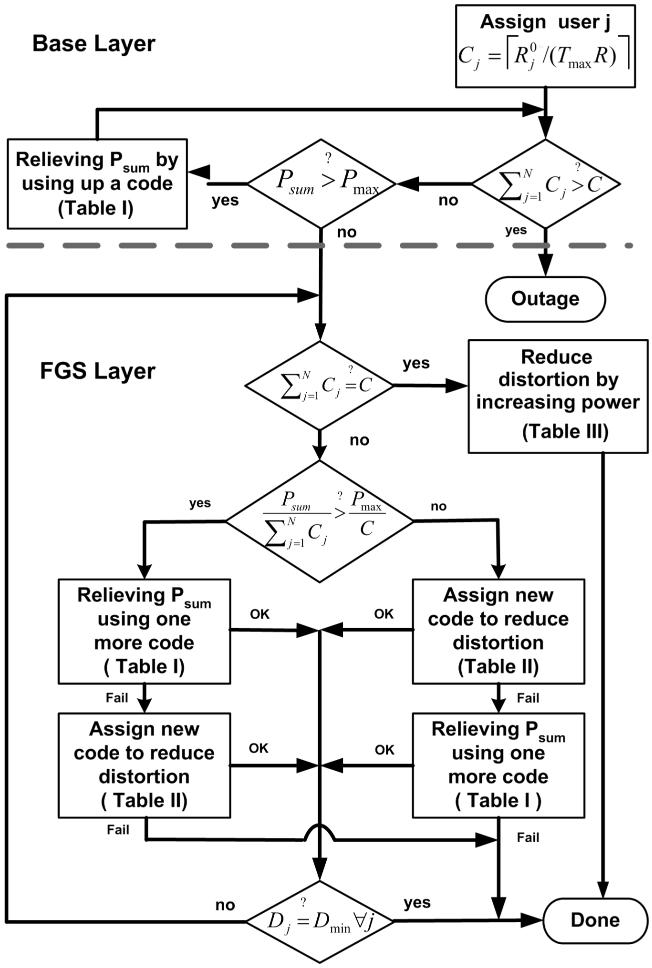

Fig. 3. Distortion Management Algorithm for base layer and FGS layer.

the power most while keeping the distortion fixed, until the power constraint is satisfied. By assigning one more code but fixing source rates, we can reduce the average channel coding rate per code. Consequently, from (3), the required SINR and power are reduced.

The power relief algorithm for P_{sum} is shown in Table I. Before an actual code is assigned to a specific user, a hypothesis is made for each user that a candidate code is assigned to this user to relieve power consumption in transmission while the settings of the other users are kept unchanged. For example, the j^{th} user will keep his/her source coding rate, r_j, unchanged, but redistribute r_j among his/her already assigned codes plus the candidate code. Subsequently, the channel coding rates for those codes, the required SINR, and the overall transmitted power are reduced. This problem can be solved using a greedy method. First, the candidate CDMA code is assigned with the maximum channel coding rate T_{max}, so the throughput is initially larger than r_j. Since Y_i in (5) is a monotonically increasing function of T_i, P_{sum} can be reduced by searching the code with the largest gradient $g_i^T = |\frac{\partial P_{sum}}{\partial T_i}|$ and then reducing the channel coding rate of this code. The algorithm repeats the above power reduction step until the throughput is equal to r_j. Finally from all hypotheses, the user reducing P_{sum} by the highest amount is selected and assigned with a real code.

TABLE I

P_{sum} RELIEF ALGORITHM

1. For hypothesis $j = 1$ to N:
   - Assign a candidate code to user j.
   - Calculate the optimal T_i for all codes assigned to user j including the candidate code, such that P_{sum} is reduced most, while r_j is unchanged.
2. Pick the user with the largest reduced P_{sum} and assign him/her a real code.

TABLE II

CODE ASSIGNMENT TO REDUCE DISTORTION

1. For hypothesis $j = 1$ to N:
   - Assign user j a candidate code, analyze ΔD_j,P_{sum}
   - If $P_{sum} < P_{max}$, add hypothesis j to candidate list.
2. If there is no candidate user, do not assign the code and go to the algorithm in Table I.
3. Among the candidates, choose the one with the largest ΔD_j and assign a real code to user j.

C. Resource Allocation for FGS Layer

After the initialization, we apply the distortion management algorithm as shown in the lower part of Figure 3 to allocate resources for FGS layer to reduce users' overall distortion. The system has two types of resources, namely, code and power. Since the distortions can be reduced by either using extra power or codes, if any type of resources is used up early than the other type, we may arrive at some undesired local optimum. So the main principle of the proposed algorithm is to fully utilize both resources by keeping a guideline to the ratio between the usage of power and codes as follows: If the current overall transmission power P_{sum} over the current number of assigned codes $\sum_{j=1}^{N} C_j$ is larger than P_{max}/C (i.e. the ratio of the maximal transmission power to the overall number of CDMA codes), the system is in a power unbalanced state by consuming higher than average power per code. A new code is assigned to relieve transmitted power usage such that the ratio of the consumed power to the number of assigned codes is reduced. Otherwise, the system is in power balanced state and a new code can be assigned to reduce the overall distortion.

More specifically, for the power unbalanced state, the algorithm in Table I, which has been discussed in Section III-B, is applied to reduce transmission power. If power cannot be relieved by the algorithm in Table I and $P_{sum} < P_{max}$, we can still improve quality by switching to the distortion reduction subsystem as shown in Table II (with more power consumption). If the system fails in reducing distortion, the algorithm terminates.

For the power balanced state, we first apply distortion reduction algorithm (Table II). The rate is increased and the overall distortion is reduced by assigning one code to a user at a time. Before an actual code is assigned to a specific user, a hypothesis is made for each user that a candidate code with channel coding rate T_{max} is assigned to the user while the settings of the other users are kept unchanged. We then use (6) to calculate the overall required power for transmission, P_{sum}, and the reduced distortion of the received video is given by:

$$\Delta D_j = D_j(r_j) - D_j(r_j + RT_{max}) \tag{8}$$

TABLE III

DISTORTION REDUCTION BY INCREASING POWER

1. For hypothesis $i = 1$ to C:
   - If T_i of code i is equal to T_{max}, do next hypothesis.
   - For code i, calculate the corresponding decrease in channel coding rate of one discrete step, ΔT_i.
   - Given ΔT_i, calculate Δr_j, ΔD_j, and ΔP_{sum}. If $P_{sum} < P_{max}$, add hypothesis i to candidate list.
2. If no candidate left, exit. Otherwise, choose the code with the largest $|\Delta D_j / \Delta P_{sum}|$ and change the channel coding rate to the chosen code.
3. Empty candidate list. Go to step 1.

where r_j is the user's current source rate. If P_{sum} is smaller than P_{max}, this hypothesis is added into a candidate list. Then among all hypotheses in the list, an actual code is assigned to the user who can reduce a highest amount of distortion. If there is no candidate in the list, it means the overall distortion cannot be further reduced. In this situation, we reduce the power using the algorithm in Table I.

When all codes are assigned (i.e. $\sum_{j=1}^{N} C_j = C$) and there is still some transmission power left, the overall distortion can be further reduced by increasing the power assignment and the channel coding rate via (3) such that more FGS data can be accommodated. To efficiently exploit the remaining power, we develop an iterative algorithm to distribute power among CDMA codes. In each iteration, we make C hypotheses to examine which code can reduce the distortion most by using the least transmission power. For code i, we check whether its channel coding rate T_i is less than T_{max}. If so, we increase T_i by a discrete step ΔT_i according to the available channel coding rates and keep the settings of the rest $C - 1$ codes unchanged. If this code belongs to the j^{th} user, the reduced distortion ΔD_j and increased overall power $\Delta P_{sum}^{(j)}$ are calculated. The hypothesis of increasing the channel coding rate of this code is added to a candidate list if $\Delta P_{sum}^{(j)} + P_{sum} \leq P_{max}$. After scanning all codes, among all candidates, our strategy is to pick the code with the largest $|\Delta D_j / \Delta P_{sum}^{(j)}|$ and set $T_i = T_i + \Delta T_i$. Then the algorithm updates the transmitted power usage, empties the candidate list, and repeats the above procedure until there is no candidate code in the candidate list. The detailed algorithm is listed in Table III.

It is worth mentioning that only the entries related to a specific code need to be updated each time when evaluating P_{sum} in (6). Thus, the overall complexity of our proposed algorithm is $O(C^2)$. This is substantially lower than the brute force search and is feasible as the base station typically can afford such a computation burden in practice.

IV. DEMAND-LIMITED, CODE-LIMITED, AND POWER-LIMITED SOLUTIONS

Because of the mixed integer programming nature of the problem formulation in (7), it is difficult to evaluate how close to the optimal solutions the proposed algorithm performs. However, we have found it possible to derive optimal solutions or a performance upper bound of (7) for three special cases, namely, demand-limited case, code-limited case, and power-limited case. In this section, we discuss these cases in details and compare in the next Section these results with the performances of the proposed scheme.

A. Demand-Limited Case

The demand-limited case refers to the situation when the number of users is small or the overall requested source rates are very low, so that both the power and code resources are abundant for the users. Under this condition, the problem (7) has only source rate constraints. Since source R-D function is a monotonically decreasing function, the optimal solution to achieve the lowest distortion is that all users transmit the maximal source rate R_j^P.

B. Code-Limited Case

The code-limited case refers to the situation when the transmission power can be viewed as unbounded, while there are only a limited number of pseudo-random codes. This happens when all users are close to the base station, so that the necessary transmission power $P_{sum} \ll P_{max}$. We also assume the number of users is large enough or the requested rates for video transmission are large enough, so that all codes are used, i.e., $\sum_{j=1}^N a_{ij} = 1, a_{ij} \in \{0,1\}, \forall i$. In this case, in order to have the highest distortion reduction, each code should carry as much information as possible. So all channel coding rate T_i should be equal to T_{max}. Under these conditions, the problem (7) becomes

$$\min_{r_j} \sum_{j=1}^N D_j(r_j) \tag{9}$$

$$\text{subject to } \begin{cases} R_j^0 \le r_j \le R_j^P, \forall j; \\ \sum_{j=1}^N r_j = CRT_{max}. \end{cases}$$

To solve the above problem, we first allocate each user with enough codes to accommodate the rate for the base layer. The number of codes that each user has is $C_j = \lceil \frac{R_j^0}{RT_{max}} \rceil$. If the total number of codes for the users to transmit their base layers is more than the total number of codes, i.e., $\sum_{j=1}^N C_j > C$, an outage is reported. Otherwise, we use the remaining codes to transmit FGS enhancement layers to further reduce distortion.

To transmit FGS layers, we perform the following procedures. First, notice that after allocating the base layer, there is an unused bandwidth of $C_j RT_{max} - R_j^0$, with the codes assigned to user j. To fully utilize the allocated bandwidth, we need to put FGS rates in these already assigned codes. Let $C' = C - \sum_{j=1}^N C_j$ be the number of unassigned codes. The next step is to distribute these C' codes for FGS layer. To facilitate our discussion, we introduce a new variable $r_j' \triangleq r_j - C_j RT_{max}$ and define a shifted version of R-D function $\tilde{D}_j(r_j') \triangleq D_j(r_j)$. Since each code carries information of RT_{max} bits, each user's unsent FGS layer bit stream can be divided into $C_j' = \lceil \frac{R_j^p - C_j RT_{max}}{RT_{max}} \rceil$ segments with equal length of RT_{max} bits. Let Δd_{jk} be the distortion reduction of the k^{th} segment of FGS layer for the j^{th} user, i.e.,

$$\Delta d_{jk} = |\tilde{D}_j(kRT_{max}) - \tilde{D}_j((k-1)RT_{max})|. \tag{10}$$

Further, let $y_{jk} = 1$ if the j^{th} user is allowed to transmit the k^{th} FGS layer segment; otherwise $y_{jk} = 0$. Notice that since the rate-distortion function of each user is convex and decreasing, the following relationship holds: $\Delta d_{jk} \ge \Delta d_{jk'}, \ \forall k' > k$. To ensure the receiver can decode all the received data of the FGS layer, the received segments should be a truncated version of the FGS bit stream such that all the received bit streams can be decoded, i.e., if $y_{jk} = 1$, then $y_{jk'} = 1, \ \forall k' < k$. The problem (9) can be reformulated as an assignment problem, which is to choose $\{y_{jk}\}$ such that the overall distortion reduction for FGS enhancement layer is maximized:

$$\max_{y_{jk}} \sum_{j=1}^N \sum_{k=1}^{C_j'} y_{jk} \Delta d_{jk} \tag{11}$$

$$\text{subject to } \sum_{j=1}^N \sum_{k=1}^{C_j'} y_{jk} = C'.$$

The optimal solution for the above problem is to sort all Δd_{jk} in a decreasing order and obtain the corresponding indices, $I(j,k) \in \{1, 2, ..., \sum_{k=1}^N C_j'\}$. For example, if $\Delta d_{j_1 k_1} \ge \Delta d_{j_2 k_2} \ge \ldots$ is the sorting result, we assign $I(j_1, k_1) = 1$, $I(j_2, k_2) = 2$, and so on so forth. Then, we pick the first C' indices and assign codes to the corresponding users for transmitting the selected FGS bit stream segments. The above solution is optimal since if we replace any item $I(j,k) \le C'$ in the above selected set with an item $I(j',k') > C'$, the resulted distortion reduction of the latter case will be smaller than the former one due to $\Delta d_{jk} > \Delta d_{j'k'}$. Note that the optimal solution also guarantees all the received bit stream of each user is a truncated version of the FGS bit stream. This is because if user j receives the k^{th} segment, user j must have received the segment 1 to $k-1$ since $\Delta d_{jk'} \ge \Delta d_{jk}, \ \forall k' < k$ or $I(j,k') < I(j,k), \ \forall k' < k$ according to the decreasing order [35].

As we can see from (11) and the simulation results later, the solution of the code-limited case is influenced by the rate-distortion functions and is not affected by the specific channel conditions. So under this condition, the video content is the dominant factor on resource allocation. The users who can reduce their distortion more with a request of smaller rates will have priority to transmit their FGS enhancement layers.

C. Power-Limited Case

The power-limited case refers to the situation that all available power is used and there might still be some codes left. This happens when all users are far away from the base station. We also assume the number of users is small or the requested rates are not large, so that it is not limited by power and code simultaneously. To simplify the analysis, we further assume that the downlink is synchronized, i.e., $\alpha_{ki} = 0, \ \forall i \ne k$. So the overall power can be expressed as

$$P_{sum} \approx \sum_{j=1}^N \sum_{i=1}^C a_{ij} \frac{\sigma^2 2^{AT_i + B} R}{WG_j}. \tag{12}$$

Since the number of available code is unlimited and P_{sum} is a convex and increasing function of T_i, using Jensen's inequality, we can draw a conclusion that the channel coding rate T_{min} for all codes will have the minimal overall transmission power. So if the power is limited, by using the minimal channel coding rate, we can have the highest source rates and corresponding minimal distortions.

Similar to the code-limited case, we first satisfy the base layer's requirement. The required number of codes is $C_j = \lceil \frac{R_j^0}{RT_{min}} \rceil$. The required transmission power for base layer is

$$P_{sum}^{base} \approx \sum_{j=1}^{N} \sum_{k=1}^{C_j} \frac{\sigma^2 2^{AT_{min}+B} R}{WG_j}. \tag{13}$$

If P_{sum}^{base} is more than P_{max}, an outage is reported; otherwise, calculate the remaining power budget $P_{max}^{budget} = P_{max} - P_{sum}^{base}$ and perform the following procedures. First, the remaining rates of the code for the base layer, $C_j RT_{min} - R_j^0$, is assigned with FGS layer rate. Then, the rest of codes are assigned for FGS layers. To facilitate our discussion, we define $r_j'' \triangleq r_j - C_j RT_{min}$ and a shifted version of R-D function $\hat{D}_j(r_j'') \triangleq D_j(r_j)$. Let P_{sum}^{FGS} be the overall required power to transmit the additional FGS layer. The reformulated problem for FGS layer is

$$\min_{r_j''} \sum_{j=1}^{N} \hat{D}_j(r_j'') \tag{14}$$

$$\text{subject to} \begin{cases} 0 \le r_j'' \le R_j^p - C_j RT_{min} \ \forall j, \\ P_{sum}^{FGS} \le P_{max}^{budget}. \end{cases}$$

Since each code will carry information of RT_{min} bits, we divide each user's unsent FGS layer bit stream into $C_j'' = \lceil \frac{R_j^p - C_j RT_{min}}{RT_{min}} \rceil$ segments with equal length, RT_{min}. Let $\Delta d_{jk}''$ be the distortion reduction for the k^{th} segment of FGS layer for the j^{th} user, i.e.,

$$\Delta d_{jk}'' = |\hat{D}_j(kRT_{min}) - \hat{D}_j((k-1)RT_{min})|. \tag{15}$$

Further, let $x_{jk} = 1$ if the j^{th} user is assigned with the k^{th} FGS layer segment and we will allocate a code to transmit this segment; otherwise $x_{jk} = 0$. Define the required power for user j if we transmit the k^{th} segment as $P_{jk} = \frac{\sigma^2 2^{AT_{min}+B} R}{WG_j}$. Since the required power for each user is only related to the path loss, $P_{jk} = P_{jk'} \ \forall k' \neq k$. We can reformulate the problem (14) as an assignment problem:

$$\max_{x_{jk}} \sum_{j=1}^{N} \sum_{k=1}^{C_j''} x_{jk} \Delta d_{jk}'' \tag{16}$$

$$\text{subject to} \begin{cases} \sum_{j=1}^{N} \sum_{k=1}^{C_j''} x_{jk} P_{jk} \le P_{max}^{budget}, \\ x_{jk} = 0 \text{ or } 1, \forall j, k. \end{cases}$$

This problem is a classical 0-1 knapsack problem [36], which is an NP hard problem. However, by allowing the continuous relaxation $0 \le x_{jk} \le 1$, the simplified continuous problem of (16) can have an optimal solution with complexity $O(\sum_{j=1}^{N} C_j'')$. We define $\Delta \bar{d}_{jk} = \frac{\Delta d_{jk}''}{P_{jk}}$ and sort $\Delta \bar{d}_{jk}$ in a decreasing order to obtain the corresponding indices $I(j,k) \in \{1, 2, ..., \sum_{k=1}^{N} C_j''\}$ and the inverse indices $J(m) \in \{(j,k)\}$. For example, if $\Delta \bar{d}_{\hat{j}\hat{k}}$ is the largest value among all $\{\Delta \bar{d}_{jk}\}$, then $I(\hat{j}, \hat{k}) = 1$ and $J(1) = (\hat{j}, \hat{k})$. We define a *critical item* M as

$$M = \min\{s : \sum_{m=1}^{s} P_{J(m)} > P_{max}^{budget}\}. \tag{17}$$

The optimal solution $\{x_{J(m)}^\star\}$ for the problem (16) is

$$x_{J(m)}^\star = \begin{cases} 1, & \text{for } m = 1, \ldots, M-1; \\ 0, & \text{for } m > M; \\ \frac{\bar{P}}{P_{J(M)}}, & \text{for } m = M; \end{cases} \tag{18}$$

where $\bar{P} = P_{max}^{budget} - \sum_{m=1}^{M-1} P_{J(m)}$.

The optimality can be proved as follows. Let $\{\bar{x}_{J(m)}\}$ be an optimal solution for the problem in (16) such that the distortion reduction is maximized. Then, the power inequality constraint of the problem in (16) is active, i.e.,

$$\sum_m \bar{x}_{J(m)} P_{J(m)} = P_{max}^{budget} = \sum_m x_{J(m)}^\star P_{J(m)}. \tag{19}$$

Since $\{x_{J(m)}^\star\}$ is also an optimal solution for the problem in (16), bringing in (18) and (19), we obtain

$$\sum_{m=1}^{M-1} (\bar{x}_{J(m)} - 1) P_{J(m)} + (\bar{x}_{J(M)} - x_{J(M)}^\star) P_{J(M)} + \sum_{m>M} \bar{x}_{J(m)} P_{J(m)} = 0. \tag{20}$$

Suppose there is a $\bar{x}_{J(\alpha)} < 1$ for some $\alpha < M$, then we must have $\bar{x}_{J(\beta)} > x_{J(\beta)}^\star$ for at least one item $\beta \ge M$ from (20). Given a sufficiently small $\varepsilon > 0$, we could increase the value of $\bar{x}_{J(\alpha)}$ by ε and decrease the value of $\bar{x}_{J(\beta)}$ by $\varepsilon P_{J(\alpha)}/P_{J(\beta)}$. However, the value of the overall distortion reduction increases by $\varepsilon \Delta d_{J(\alpha)}'' - \varepsilon \Delta d_{J(\beta)}'' P_{J(\alpha)}/P_{J(\beta)}$, which is larger than 0 since $\Delta d_{J(\alpha)}''/P_{J(\alpha)} > \Delta d_{J(\beta)}''/P_{J(\beta)}$. This is a contradiction to $\{\bar{x}_{J(m)}\}$ being the *optimal* solution. Therefore, we have shown that $\bar{x}_{J(\alpha)} < 1$ for $\alpha < M$ and $\bar{x}_{J(\beta)} > 0$ for $\beta > M$ are impossible. Consequently, $x_{J(M)}^\star = \frac{\bar{P}}{P_{J(M)}}$.

Because the cross correlation between codes is assumed to be zero and the continuous relaxation for x_{jk}, the above solution provides an upper bound for the power-limited case. With the same reason for the code-limited case, the above solution guarantees that all the received bit stream of each user is the truncated version of the original FGS bit stream and can be decoded due to the decreasing sorting order.

From the problem formulation in (16) and the simulation results shown later, the solution of the power-limited case is jointly influenced by the channel conditions and video contents. The users closer to the base station and having simpler video content complexity will dominate the resource use. In reality, users may not be distributed all close to or all far away from the base station. In other words, the system is mostly constrained by both the number of available codes and the available transmission power. Our simulation results presented in the next section will show how well the proposed scheme in Section III performs for these special cases.

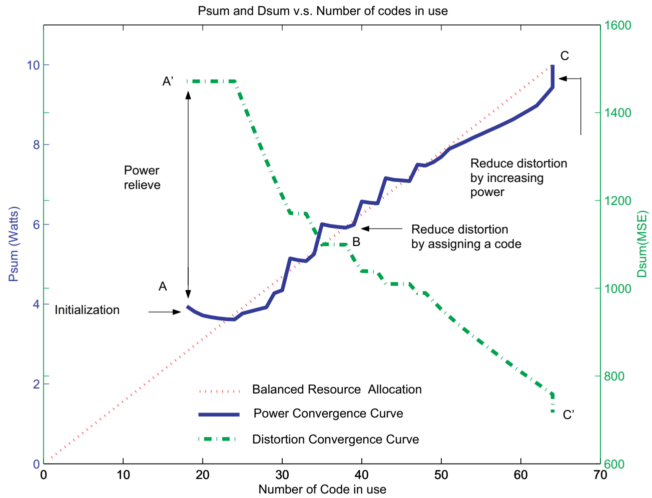

Fig. 4. Power and distortion vs. the number of assigned codes.

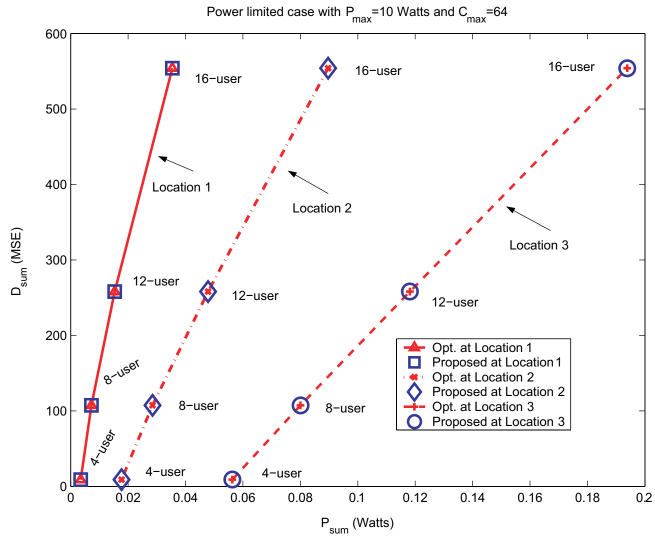

Fig. 5. Code limited case: optimal solutions.

V. Simulation Results

In order to evaluate the performances of the proposed scheme, we conduct simulations with a setup as follows. For the MC-CDMA system, the total bandwidth, W, is 7.5 MHz and the spreading factor is 64. The channel condition is stable within each video frame and changes according to Rayleigh Fading over different frames. The propagation path loss factor is 4. We use the average orthogonality factor in [33]. The noise power is 10^{-11} Watts, and the maximal transmission power is 10 Watts. For the channel coding, we use RCPC codes with a memory 4, puncturing period 8, and mother code rate $1/4$ [29]. The range of channel coding rate is set to $[T_{min}, T_{max}]$ = $[1/4, 1/2]$. Without loss of generality, we assume the base layer and the enhanced layer have the same BER requirements. Our experimental results show that to achieve BER $= 10^{-6}$ using FGS codec (including base and FGS layer), parameters (A, B) in (3) are (4.4, -1.4). For source coding, we use a MPEG-4 FGS codec with error resiliency and concealment. Error resiliency is implemented by inserting re-synchronization markers in each frame's bitstream. Error concealment is implemented by replacing lost blocks by the motion compensated corresponding blocks in the previous frame, and by replacing lost motion vectors with zero values. For the video source, we concatenate 15 classic QCIF (176 $\times$ 144) video sequences with temporal down sampling factor 2 to form a basic testing video sequence source of 2775 frames with a video refresh rate 15 frames per second. The 15 sequences are 150-frame *Akiyo*, 75-frame *Carphone*, 240-frame *Claire*, 150-frame *Coastguard*, 150-frame *Container*, 195-frame *Foreman*, 435-frame *Grandmother*, 165-frame *Hall objects*, 75-frame *Miss American*, 480-frame *Mother and daughter*, 150-frame *MPEG4 news*, 210-frame *Salesman*, 150-frame *Silent*, 75-frame *Suzie*, and 75-frame *Trevor*. The base layer is generated by MPEG-4 encoder with a fixed quantization step of 30 and a GOP pattern of 14 P frames after each I frame. All frames of FGS layer have up to six bit planes.

A. Convergence Track of Proposed Algorithm

We use Figure 4 to illustrate how the proposed scheme balances the code and power limitation to fully utilize the system resources, where we can see the convergence track of the overall power and video distortion with respect to the number of assigned codes by using the proposed algorithm. After initialization (shown at position A), a total of 18 codes are assigned to deliver the base layer of all 16 users. The overall visual distortion (shown at position A') is large because only the base layer is transmitted. We can see that at this point, the system is power unbalanced, i.e. the operating point A is above the balanced resource allocation line. Whenever the power unbalance occurs, we apply the power relief algorithm in Table I to reduce the power while keeping the distortion fixed. When the system is not power unbalanced (such as position B), we assign codes to reduce distortion using the algorithm in Table II, until all the codes are used up. Note that by doing so, the required power is increased. Finally, we use the algorithm in Table III to further reduce distortion (shown at position C') by the remaining power quota (shown at position C). At the end, all available power and code resources are fully utilized.

B. Performance under Special Cases

To study the proposed algorithm performance under code-limited (power-unlimited) case we compared its total distortion D_{sum} ($D_{sum} = \sum_{j=1}^{N} D_j$) with that of the optimal solution of the code-limited case for different number of users, N, and different locations. We set all mobile users at locations 1, 2, and 3, at distances near 100m, 150m, and 200m, respectively. From the results shown in Figure 5, we can see that the proposed scheme always achieves the optimal solution.

To study the proposed algorithm performance under power-limited (code-unlimited) case, we considered $N = 4$ users which are located at different locations. All mobile users in location 1,2 and 3, are near 1100 m, 1200 m, and 1300 m, respectively. Figure 6 shows the results comparing the proposed algorithm with orthogonality factor between 0 and 0.7 and the performance upper bound for the power-limited case. As we can see, the performance of the proposed algorithm with small orthogonality factor is close to the performance upper

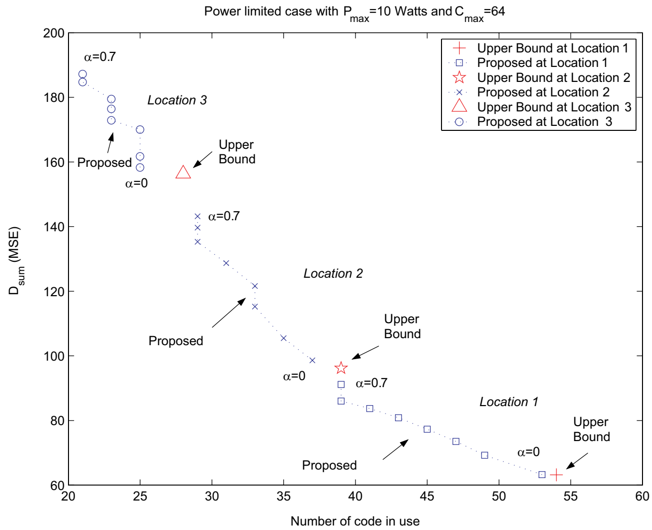

Fig. 6. Power limited case: close to performance upper bounds.

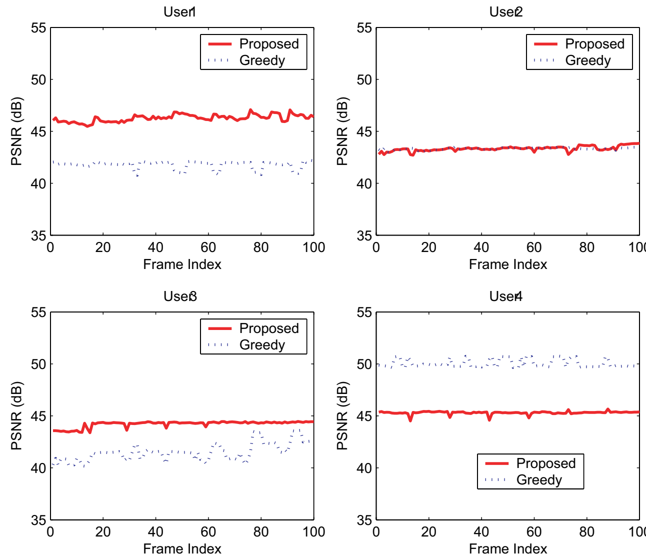

Fig. 7. Frame-by-frame PSNR results for user 1 to user 4 vs. frame index.

bound that assumes the orthogonality factor to be zero. Furthermore, we measure the relative D_{sum} difference between the proposed algorithm with zero orthogonality factor and the performance upper bound, the average performance loss for 100 frames is only 2.25%. The loss is because two reasons. First, the performance upper bound is obtained by allowing a non-integer number of codes, so the bound has a better performance than the optimal solution of (7). Second, the proposed algorithm might reach local minima instead of the global minima. The above two simulation results demonstrate the effectiveness of the proposed algorithm in both special cases.

C. Performance Results in General Scenario

For the case where both power and code are constrained, we compare the proposed algorithm with a modified greedy approach [30]. This modified approach is similar to our proposed framework, but uses a greedy approach for the code assignment in FGS layer. For the base layer, the greedy algorithm executes the same procedure as our proposed algorithm. For each iteration in FGS layer, this greedy algorithm tries to assign a candidate code with channel coding rate, T_{max}, to every user, calculates $|\Delta D_j/\Delta P_{sum}|$, and assigns a new code to the user with the largest value. This greedy scheme will favor the users close to the base station and with simple video content complexity.

Figure 7 shows the frame-by-frame PSNR results in a four-user system in which user $1 \sim 4$ are located at 700m, 400m, 600m, and 20m, respectively. User 1 to 4 receive 100-frame video sequence of *Claire*, *Coastguard*, *Grandmother*, and *Akiyo*, respectively. The first three users receive better or similar video qualities using the proposed algorithm. The greedy algorithm assigns codes to the users who can use the least power to obtain the largest decreased distortion, which is the fourth user in this example. Compared with the proposed scheme, the greedy scheme cannot effectively reduce other three users' distortions.

Figure 8 shows the number of users v.s. the average of the total distortion D_{sum}. The content program for each user is 100 frames and starts from a randomly selected frame of the concatenated testing video. The location for each user is uniformly distributed within the cell with radius from 20 m to 1000m. We repeat the simulations 300 times. The simulation results demonstrate that the average D_{sum} of the proposed algorithm outperforms that of the greedy algorithm $14\% \sim 26\%$. The 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. In other words, only one system constraint in (7) becomes active after several iterations, which leaves no room to improve the overall system performance even if other resources are still available. The proposed scheme shows performance improvement by fully utilizing both power and code resources.

VI. Conclusions

In this paper, we propose a framework for transmitting multiple video streams over wireless communication networks, which is a promising service over the current wireless networks. Specifically, we have developed a system to transmit multiple real-time MPEG-4 FGS video programs over downlink multicode CDMA networks. The resource allocation is formulated as an optimization problem to minimize the overall received distortion of all users subject to the baseline video quality requirement, the maximal transmission power, and the number of codes constraints. To fully utilize the limited resources, we propose a distortion management algorithm to jointly allocate source coding rates, channel coding rates, CDMA codes, and transmission power. The scheme balances the constraints of codes and power so that no resources are wasted. We also derive optimal solutions for the demand-limited and code-limited case and a performance upper bound for the power-limited case.

Experimental results show that the proposed approach provides an efficient solution for sending videos over downlink MC-CDMA system. The system can fully utilize the available radio resources by balancing the power and code limits. Regarding the three special cases, the proposed scheme

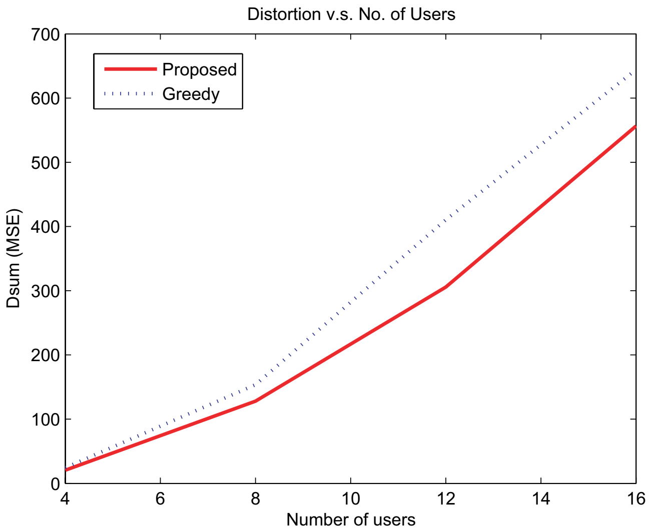

Fig. 8. Performance comparison of the proposed and greedy schemes.

can achieve optimal solutions in demand-limited and code-limited cases, and can have less than 2.25% performance loss compared to the performance bound for the power-limited case. Compared with a greedy algorithm in the literature, the proposed algorithm can outperform by 14% to 26%. The proposed scheme is a promising solution for real-time multiple video transmissions in current and future CDMA networks.

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Zhu Han (S'01-M'04) received the B.S. degree in electronic engineering from Tsinghua University, in 1997, and the M.S. and Ph.D. degrees in electrical engineering from the University of Maryland, College Park, in 1997 and 2003, respectively.

From 2000 to 2002, he was an Engineer in the R&D Group of ACTERNA, Maryland. From 2002 to 2003, he was a Graduate Research Assistant at the University of Maryland. From 2003 to 2006, he was a Research Associate at the University of Maryland. Currently He is an assistant Professor in Electrical and Computer Engineering Department at Boise State University, Idaho, USA. His research interests include wireless resource allocation and management, wireless communications and networking, game theory, and wireless multimedia.

Dr. Han is a member of the Technical Programming Committee for the IEEE International Conference on Communications of 2004, 2005, and 2007, the IEEE Vehicular Technology Conference, Spring 2004, the IEEE Consumer Communications and Networking Conference 2005, 2006, and 2007, the IEEE Wireless Communications and Networking Conference 2005, 2006 and 2007, and the IEEE Globe Communication Conference 2005 and 2006, as well as Session Chair of the IEEE Wireless Communications and Networking Conference 2004, 2005, 2006 and the IEEE Globe Communication Conference 2005.

Guan-Ming Su (S'04) received the B.S.E. degree from National Taiwan University, Taipei, Taiwan, in 1996 and the M.S. degree from University of Maryland, College Park, in 2001, both in electrical engineering. He is currently working toward the Ph.D. degree in the Department of Electrical and Computer Engineering at the University of Maryland, College Park.

He was with the Research and Development Department, Qualcomm, Inc, San Diego, CA, during the summer of 2005, and is currently with ESS Technology, Inc., Fremont, CA. His research interests are multimedia communications and multimedia signal processing.

Andres Kwasinski received in 1992 his diploma in Electrical Engineering from the Buenos Aires Institute of Technology, Buenos Aires, Argentina, and the MS and PhD degrees in Electrical and Computer Engineering from the University of Maryland, College Park, Maryland, in 2000 and 2004, respectively. Since 2004 he has been a Faculty Research Associate in the Department of Electrical and Computer Engineering at the University of Maryland, working on wireless multimedia communications. In addition, he worked during 1993 for NEC as telephone switches software developer and from 1994 to 1998 for Lucent Technologies in several capacities. His research interests are in the area of multimedia wireless communications, multimedia cooperative communications, cross layer designs, and speech and video processing (especially in the area of signal compression). Currently he is with the Texas Instrument's Advanced Signal Processing group in Germantown, Maryland.

Min Wu (S'95-M'01) received the B.E. degree in electrical engineering and the B.A. degree in economics (both with the highest honors) from Tsinghua University, Beijing, China, in 1996, and the Ph.D. degree in electrical engineering from Princeton University in 2001.

Since 2001, she has been on the faculty of the Department of Electrical and Computer Engineering and the Institute of Advanced Computer Studies at the University of Maryland, College Park, where she is currently an Associate Professor. Previously she was with the NEC Research Institute and Panasonic Laboratories. She co-authored two books and holds five U.S. patents. Her research interests include information security and forensics, multimedia signal processing, and multimedia communications.

Dr. Wu received an NSF CAREER award in 2002, a University of Maryland George Corcoran Education Award in 2003, an MIT Technology Review's TR100 Young Innovator Award in 2004, and an ONR Young Investigator Award in 2005. She is a co-recipient of the 2004 EURASIP Best Paper Award and the 2005 IEEE Signal Processing Society Best Paper Award. She is an Associate Editor of *IEEE Signal Processing Letters*, and served as a Guest Editor of a 2004 special issue in *EURASIP Journal on Applied Signal Processing* and Publicity Chair of 2003 IEEE International Conference on Multimedia and Expo.

K. J. Ray Liu (F'03) received the B.S. degree from the National Taiwan University in 1983, and the Ph.D. degree from UCLA in 1990, both in electrical engineering. He is Professor, Associate Chair and Director of Graduate Studies and Research of Electrical and Computer Engineering Department, University of Maryland, College Park. His research contributions encompass broad aspects of wireless communications and networking, information forensics and security, multimedia communications and signal processing, bioinformatics and biomedical imaging, and signal processing algorithms and architectures.

Dr. Liu is the recipient of numerous honors and awards including best paper awards from IEEE Signal Processing Society (twice), IEEE Vehicular Technology Society, and EURASIP; IEEE Signal Processing Society Distinguished Lecturer, EURASIP Meritorious Service Award, and National Science Foundation Young Investigator Award. He also received Poole and Kent Company Senior Faculty Teaching Award and Invention of the Year Award, both from University of Maryland.

Dr. Liu is Vice President - Publications and on the Board of Governor of IEEE Signal Processing Society. He was the Editor-in-Chief of *IEEE Signal Processing Magazine* and the founding Editor-in-Chief of *EURASIP Journal on Applied Signal Processing*.