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Joint Call Admission Control and Resource Allocation for H.264 SVC Transmission over OFDMA Networks

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Abstract— **This paper aims to combine adaptive subcarrier allocation and bit loading with the transmission of the H.264 SVC (Scalable Video Coding) encoded video sequences in order to increase the number of supported users in the system and provide the best quality of service (QoS) to the subscribers. We initially assume that the number of calls at the base station can be supported, and present an integer program (IP) formulation of the problem that considers the frequency selective nature of the channel, bit error rate requirement and the discrete rate requirements of the different layers of the medium grain scalable (MGS) video. It is shown how the IP can be extended to perform call admission control (CAC). Due to the complexity involved with IP, a sub-optimal scheme is then presented. Results demonstrate that our proposed scheme performs better than systems with a fixed resource allocation strategy by supporting more users and by always achieving acceptable QoS. Furthermore, the low complexity of the proposed CAC schemes makes it suitable for practical application.**

I. INTRODUCTION

Due to the emergence of broadband networks and the desire to support high data rates, there has been a tremendous increase in the demand for multimedia delivery over the wireless medium in recent years. This trend is also partly due to the advancement of video compression standards, such H.264/AVC [1], that achieve high compression efficiency and have improved reliability. Nonetheless, the error-prone and interference limited nature of the wireless environment hinders the deployment of multimedia services. Multipath fading and ISI lead to drastic variations in channel capacity and channel error rate, both of which are unfavorable for multimedia communication.

Numerous techniques have been proposed to cater for channel variations and packet losses [2], [3]. One such method is scalable video coding (SVC), i.e the coding of the video sequences into a single bitstream, but consisting of numerous layers, all of which need not be present for decoding at the receiver. The concept of scalable video has been realised in numerous earlier video standards, namely MPEG-2, H.263 and MPEG-4 Visual. Recently, the H.264/AVC standard has been extended to support three types of scalabilities [4]: temporal, spatial and SNR. Compared to previous scalable video coding techniques, H.264 SVC has a good compression efficiency and manageable complexity. In this work, we adopt the medium grain scalability of H.264 SVC to encode each video sequence into one mandatory base layer and one or more enhancement layers in multiuser networks.

We consider downlink transmission over an orthogonal frequency division multiple access (OFDMA) network. Resource allocation for multiuser orthogonal frequency division multiplexing (OFDM) is a subject that has been extensively studied [5]–[7]. However, the incompatibility between the network and the QoS requirement limits the direct application of such techniques. A number of cross-layer approaches for multimedia transmission have been proposed so far, [8] and references therein, with the main objective of improving QoS. In modern wireless networks however, the number of users transmitting may vary. A network may be required to allow new connections while maintaining the minimum quality requirements of the remaining users. Call admission control (CAC) is one of the most important mechanisms in modern networks, whose goal is to provide guaranteed QoS to the established calls. CAC for video transmission has been investigated in [9], [10]. To the authors' knowledge, however, no prior work has considered CAC and adaptive resource allocation jointly for transmission of scalable video.

The two novel contributions of this paper are:
- Formulation of resource allocation for scalable video transmission in an integer program which considers the rate requirements of different layers
- A CAC scheme that jointly considers resource allocation for improving the QoS of established calls

This paper is structured as follows. In section II, we define the problem assuming that there are enough resources to support all users. A standardised IP formulation is then given. Section III illustrates the proposed call admission control technique and bit loading strategy. Simulation results are shown and discussed in section IV. Finally section V concludes the paper.

Notation: We use bold uppercase (lowercase) letters to represent matrices (column vectors); $\mathbf{1}_M$ is a vector of 1's of dimension $M \times 1$ and $\mathbf{0}_{M \times N}$ is the all zero matrix of dimension $M \times N$; $\mathbf{I}_M$ is the $M \times M$ identity matrix; $\otimes$ represent the Kronecker product and $(\cdot)^T$ denotes the transpose operation.
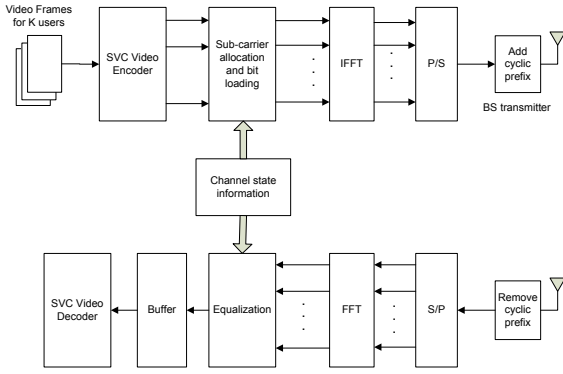
II. RESOURCE ALLOCATION FOR VIDEO TRANSMISSION

Fig. 1. Transmitter/receiver block diagram

We first address the problem of resource allocation for video transmission. Consider a downlink scenario where video data is transmitted to K subscribers using OFDM modulation. The block diagram of the system is shown in Fig. 1. It is initially assumed that the number of calls can be supported by the network. The objective is to provide an algorithm that, for a given number of calls, gives, on average, the best visual quality by intelligent allocation of subcarriers and bit loading. This is similar to the capacity maximization problem. Assuming K users and N subcarriers, the resource allocation problem can be formulated as

$$\text{max.} \quad \sum_{n=1}^{N}\sum_{k=1}^{K} c_{k,n}w_{k,n}$$

$$\text{s.t.} \quad \sum_{n=1}^{N} w_{k,n}c_{k,n} \geq r_{min}^{k}, \forall k$$

$$\sum_{k=1}^{K} w_{k,n} \leq 1, \forall n$$

$$\sum_{n=1}^{N}\sum_{k=1}^{K} \frac{f_k(c_{k,n})}{|h_{k,n}|^2} w_{k,n} \leq P_T$$

$$w_{k,n} \in \{0,1\}, \qquad c_{k,n} \in \{0, \cdots, M-1\} \quad (1)$$

where $c_{k,n}$ is the number of bits assigned to the nth subcarrier of user k, $|h_{k,n}|^2$ is the channel gain of user k on the nth subcarrier, P_T is the total power assigned for transmission and the function $f_k(c_{k,n})$ determines the minimum received power required for reliable reception of $c_{k,n}$ bits if the channel gain is unity. The first constraint ensures that at least the base layer is transmitted for all the users, with r_{min}^k corresponding to the required number of bits in each OFDM symbol to transmit the base layer. The variable $w_{k,n}$ is defined as

$$w_{k,n} = \begin{cases} 0 & \text{if } c_{k,n} = 0 \\ 1 & \text{otherwise} \end{cases} \quad (2)$$

The function $f_k(c)$ is dependent on the modulation and coding scheme supported. For uncoded M-QAM modulation and

tolerable bit error rate of P_e[1],

$$f_k(c) = \frac{N_0}{3}\left[Q^{-1}\left(\frac{P_e}{4}\right)\right]^2 (2^c - 1) \quad (3)$$

where N_0 is the noise variance and c is the number of bits per QAM symbol [5]. Alternately, a look-up table for different modulation and coding schemes can be used. To provide higher priority to the base layer, it is possible to assign a lower P_e requirement to that layer. For the uncoded case, this would imply assigning higher power for the transmission of the base layer. This approach has been studied in a number of research papers and the potentials of such technologies are well-known [11].

Solving the above optimization problem however does not guarantee an improvement in the quality of the scalable video. As opposed to the fine grain scalability (FGS) present in MPEG-4, the MGS option in the H.264 standard requires that each video layer has a discrete rate. Neglecting this fact would lead to an inefficient and sub-optimal solution. Transmitting at a rate between two video levels offers no gain compared to transmitting at the rate corresponding to the lower video layer. We note that although the optimisation in (1) is over integers, the cost function is non-linear and thus does not lend itself to solution by efficient, known algorithms. To remedy this issue, we reformulate the problem.

To cater for the discrete rates supported in the system, the first constraint in (1) must be modified as follows:

$$\sum_{n=1}^{N} w_{k,n}c_{k,n} = z_{k,1}r_1 + \cdots + z_{k,L}r_L, \forall k \quad (4)$$

$$\sum_{i=1}^{L} z_{k,i} = 1, \forall k \quad (5)$$

where r_l is the number of bits necessary in each OFDM symbol for transmitting the lth video layer and $z_{k,l}$ equals to 1 only if user k is transmitting the lth video layer. For simplicity, we can assume that all video sequences have been encoded in the same number of layers and have the same rate requirements. The relaxation of this assumption to other cases is straightforward.

Suppose that $c_{k,n} \in \{0,1,\cdots, M-1\}$ and define the new indicator variable $\rho_{k,n,c}$ as:

$$\rho_{k,n,c} = \begin{cases} 1, & \text{if } c_{k,n} = c \\ 0, & \text{otherwise} \end{cases} \quad (6)$$

Defining the vector $\mathbf{d} = [\mathbf{1}_{NK} \otimes \mathbf{c}, \mathbf{0}_{KL\times 1}]^T$ where $\mathbf{c} = [0,1,\cdots,M-1]^T$ and L being the maximum number of layers in each video stream, and the vector of indicator variable $\mathbf{x}$ as

$$\mathbf{x} = [\mathbf{y}, \mathbf{z}]^T \quad (7)$$

$$\mathbf{y} \triangleq [\rho_{1,1,1}, \rho_{1,1,2}, \cdots, \rho_{K,N,M}]^T$$

$$\mathbf{z} \triangleq [z_{1,1}, z_{1,2}, \cdots, z_{1,l}, z_{2,1}, \cdots, z_{K,L}]^T$$

[1]It should be noted that video data is very sensitive to bit errors, and P_e of the order of 10^{-10} is not uncommon

the objective function can be stated as

$$\text{maximise } \mathbf{d}^T \mathbf{x} \tag{8}$$

In vector notation, the constraints (4) and (5) become

$$\mathbf{A}_1 \mathbf{x} = \mathbf{0}_{K \times 1}, \quad \mathbf{A}_2 \mathbf{x} = \mathbf{1}_K \tag{9}$$

where $\mathbf{A}_1 \triangleq \left[\mathbf{I}_K \otimes (\mathbf{1}_N \otimes \mathbf{d})^T, -\mathbf{I}_K \otimes \mathbf{r} \right]$ and $\mathbf{A}_2 \triangleq [\mathbf{0}_{K \times KNM}, \mathbf{I}_K \otimes \mathbf{1}_L]$, and $\mathbf{r}$ is the length L vector containing the rate requirement for the different video layers. The constraint that no subcarriers can be shared by more than one user is formulated as

$$\mathbf{A}_3 \mathbf{x} \leq \mathbf{1}_N, \qquad \mathbf{A}_3 \triangleq \left[(\mathbf{1}_K \otimes \mathbf{I}_N \otimes \mathbf{1}_M)^T, \mathbf{0}_{N \times KL} \right] \tag{10}$$

With the new indicator variable ρ, the constraint pertaining to the power limit is reformulated as

$$\sum_{n=1}^{N} \sum_{k=1}^{K} \sum_{c=0}^{M-1} \frac{f_k(c)}{|h_{k,n}|^2} \rho_{k,n,c} \leq P_T \tag{11}$$

where f_k can be precalculated for different values of c, e.g., using (3). Letting $\mathbf{f}_k = [f_k(0), f_k(1), \cdots, f_k(M-1)]^T$, the vector representation of (11) can be expressed as

$$\mathbf{b}^T \mathbf{x} \leq P_T \tag{12}$$

where $\mathbf{b} = [\mathbf{g}_1 \otimes \mathbf{f}_1, \mathbf{g}_2 \otimes \mathbf{f}_2, \cdots, \mathbf{g}_K \otimes \mathbf{f}_K, \mathbf{0}_{KL \times 1}]^T$ and $\mathbf{g}_k$ is a $K \times 1$ vector with ith entry $1/|h_{k,i}|^2$.

The above problem can be solved using any integer programming methods. One commonly used technique is the *branch-and-bound algorithm* [12]. However the complexity for obtaining the optimal solution with this method increases exponentially with the number of variables. If the matrix $\mathbf{A}$, where $\mathbf{A} = [\mathbf{A}_1; \mathbf{A}_2; \mathbf{A}_3; \mathbf{b}^T]$, can be shown to be totally unimodular, i.e., all square sub-matrices have determinant ± 1 or 0, a linear relaxation of the problem would still lead to the optimal solution. However, the vector $\mathbf{b}$ can take non-integer values which makes it very difficult to achieve the total unimodularity.

The integer programming problem mentioned leads to the optimal solution provided that the problem is feasible. If the number of calls cannot be supported under the given channel conditions, the IP would return an infeasible solution. One possibility is to drop the user with the worst average-channel-gain-to-base-layer-rate ratio and solve the integer program again. This could be performed until a feasible solution is obtained leading to a highly computationally intensive operation. Such an approach is not practical for system with large variables. In the next section, we present a suboptimal resource allocation algorithm in conjunction with a novel CAC methodology in order to improve the QoS of users allowed into the system.

III. CALL ADMISSION CONTROL

In practical situations, it is fair to expect that the number of calls established by the base station would vary. Mobile subscribers may leave a cell, whereby more resources would become available to the remaining users, or a new call would need to be established. In the latter case, the base station must ensure that the QoS targets of all users are satisfied. Also, changes in the underlying channel cause the channel capacity to fluctuate. Users who at a given time might receive all video layers may no longer be able to in the future. The objective in almost all mobile networks is to support the maximum number of subscribers at a given time within the power budget and under the channel conditions. This is performed by a call admission control (CAC) mechanism. In this section, we consider the case where the number of calls exceed the maximum allowable limit and demonstrate how the base station can cope with the demand for wireless resources. We next propose a two-step CAC scheme that estimates the maximum number of users in the system and then performs adaptive bit loading for quality improvement.

A. Proposed CAC Algorithm

The purpose of CAC is to ensure that the users allowed into the system have a guaranteed QoS, which for SVC media transmission corresponds to receiving the base layer. In the proposed algorithm, we initially assume that a constant modulation scheme is employed for all users across all subcarriers, say BPSK modulation. Based on the fact that the base layer rate is known and a constant modulation scheme is used, it is possible to calculate the number of subcarriers required by kth user, which we denote n_k. Since the total number of subcarriers is fixed and known, we can set a lower limit to the number of calls supported, K. If the initial number of calls exceeds that amount, we refuse calls to those users whose average channel gain to base layer rate ratios are the lowest until K calls are left.

For the remaining K calls, the following novel CAC algorithm is performed to ensure that these users can be serviced within the power budget. The procedure is shown in Table I.

TABLE I

CALL ADMISSION CONTROL

1. Arrange users in descending order according to their average channel gain
2. Set $p_{used} = 0$, where p_{used} is the power required for transmission
3. Set $k = 1$
4. Calculate the power required to transmit on the n_k best subcarriers of user k
5. Update p_{used}

6. **if** $p_{used} > P_T$
      $k = k - 1$
      Stop, maximum number of calls supported is $K^* = k$
      Exit algorithm

7. **else**
      $k = k + 1$
      Remove subcarriers used by user k, from further allocation

8. Go to step **4**.

In this greedy approach, we start by allocating the subcarriers to the user with the best average channel gain. Since the users with the best subcarriers tend to require less power, and more power is assigned to the weaker subcarriers, we ensure a certain level of fairness in the system.

Following the above algorithm, we can guarantee the minimum QoS to K^* users. The IP presented in section II can be run again for these users to obtain the most appropriate resource allocation strategy. Alternately, the heuristic for bit loading and video quality maximisation proposed next can be employed at a lower complexity. Note that although we assume that all calls are admitted together, the presented CAC can be readily extended to system with sequential admission. For each new call, the resource allocation is performed. A new call is established only if doing so would not require dropping previously established calls.

B. Suboptimal Bit Loading

After the CAC scheme presented above is executed, K^* users are allowed into the system with acceptable video quality. By adaptively bit loading the subcarriers previously assigned during call admission, it might be possible to transmit one or more enhancement layer data if doing so does not lead to a violation of the power constraint. The procedure is shown below. Let $\mathcal{S}_k$ be the set of subcarrier indexes assigned to user k and $p_{\backslash k}$ be the power required for transmitting video sequences of all users, except the kth user.

TABLE II

ADAPTIVE BIT LOADING

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1. initialisation: $k = 1$, $l = 2$
2. Evaluate $p_{\backslash k}$
3. Adaptively load bits on subcarriers in $\mathcal{S}_k$
    until layer l can be transmitted
4. Calculate additional power necessary for transmitting layer l
    of user k, p_{add}
5. if $p_{add} + p_{\backslash k} > P_T$
    Stop and exit algorithm
6. else
    if $k = K^*$
        $l = l + 1$
        $k = 1$
        Go to step **2**
    else
        $k = k + 1$
        Go to step **2**

The adaptive bit loading mentioned in step 3 can be performed according to, e.g., [5], where extra bits are loaded onto those subcarriers requiring the least additional power until the target rate is attained. In the above procedure, an extra enhancement layer is transmitted if doing so does not cause the power to exceed the predetermined limit. To ensure fairness, we prevent the user with the highest channel condition from transmitting video layer $l+1$ unless all other users can transmit layer l.

IV. RESULTS

We first compare the number of calls supported by the optimal scheme with the proposed scheme. For illustration purposes, simple systems with $N = 8$ and $N = 16$ subcarriers are considered with a base layer rate $r_{min} = 3$. Fixed QPSK modulation is assumed for the first step of the algorithm. Results averaged over 50 runs are shown in Table III.

TABLE III

AVERAGE NUMBER OF SUPPORTED USERS

Approach	$N = 8$	$N = 16$
Optimal	3.25	8.0
Proposed CAC	2.214	5.8

It is observed that for small systems, the average number of users supported under both techniques are quite close. However, as the number of subcarriers increases, the IP technique results in a higher number of supported users. For practical situations however, the complexity involved in solving the IP for a large system makes this an unattractive solution.

We next analyse the performance of the proposed CAC for scalable video transmission. To provide a benchmark for comparison, we first simulate a system where subcarriers are allocated randomly and a constant modulation scheme is used, which we refer to as 'fixed RA'. A downlink scenario with 256 subcarriers and 6 users is considered, where each user is assigned 42 subcarriers. The reference SVC video sequences, 'bus' and 'city' available on [13], in CIF format and encoded at a frame rate of 30 fps are used. Each sequence is encoded into 2 spatial layers using the JSVM (Joint Scalable Video Model) reference software [14] and transmitted over randomly generated exponentially decaying channels with 8 taps and a decay factor of $\alpha = 0.86$.

Figs. 2 and 3 compare the peak signal to noise ratio of the luminance component (Y-PSNR) values at the transmitter and receiver for the bus and city sequences with this fixed allocation scheme. On average, the city sequence was assigned better subcarriers than the bus sequence. The results are for an uncoded transmission scenario using QPSK modulation and a channel-to-noise ratio (CNR) of 22 dB. Note that Y-PSNR values below 7 dB correspond to undecodable frames.
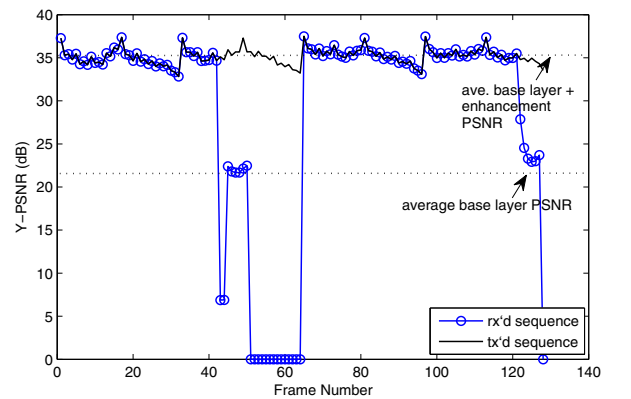

Fig. 2. Received Y-PSNR for 'city' sequence with fixed RA

The received quality of clip 'bus', Fig. 3, is considerably different from the transmitted one. With a fixed subcarrier allocation and bit loading, the transmitter can ensure that the number of bits corresponding to both base and enhancement layers are transmitted. Nonetheless, the underlying channel
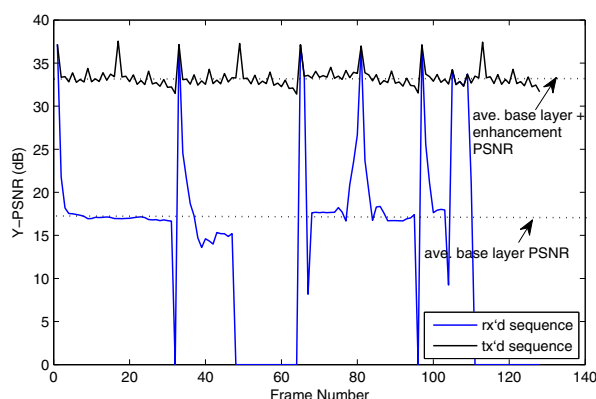
Fig. 3. Received Y-PSNR for 'bus' sequence with fixed RA

conditions are not considered. Transmitting over channels in deep fades is likely to result in decoding errors. The Y-PSNR value in the vicinity of 17 dB in Fig. 3 indicates that only the base layer has been decoded due to erroneous higher layer data. It should be noted that no ARQ is implemented in this system. Only two spatial layers were considered in this system for the purpose of illustration. In practice more graceful degradation would be observed by having more scalable layers.

The simulation was performed again, with the same channel conditions and simulation parameters, using the proposed CAC technique. Fig. 4 shows the Y-PSNR plot for the 'bus' sequence when the enhancement layer data is transmitted as well. A target bit error rate of 10^{-8} is set and the CNR fixed at 18 dB. Under such conditions, the sequence is received with virtually no errors as illustrated in the figure, even at a lower CNR. A lower CNR was used to demonstrate the superiority of the proposed scheme compared to the fixed RA. Note that
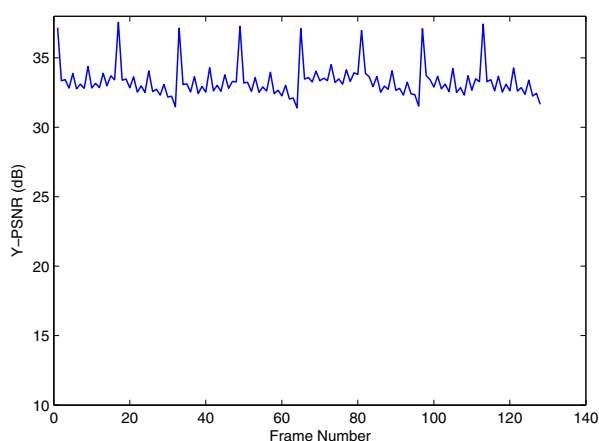

Fig. 4. Received Y-PSNR of 'bus' clip with the proposed CAC

if the maximum number of users were allowed into the system and the remaining power were insufficient, all the users would transmit only the base layer data. Also, encoding the video sequences into more than 2 layers can improve QoS over all

users with the proposed CAC due to the difference between the rates of the different video layers being smaller. Thus the extra power required for transmitting another video layer is lower. Nonetheless, more video layers would imply a larger number of variables in the IP which dramatically increases complexity.

V. CONCLUSION

An IP formulation for jointly allocating subcarriers and bit loading for transmission of SVC in OFDMA networks has been proposed. Due to the complexity involved in the IP, a sub-optimal CAC was also presented, whereby subcarriers are assigned following a greedy approach but where fairness is considered by allocating more power to the users with poor subchannels. Simulation results demonstrate that for small systems, the number of users supported by the proposed CAC is close to that of the IP-based method.

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