Abstract — Multicasting video over wireless networks is a best effort service. To multicast a video, Scalable Video Coding with spatial, temporal and quality scalabilities is adopted. Scalable Video multicast system consists of channel probing stage and transmission stage. The optimal resource allocation problem is formulated by maximizing the video quality of the clients subject to transmission energy and channel access constraints. This problem is a joint optimization of the selection of Modulation and Coding Scheme (MCS), and the transmission power allocation. The proposed scheme is of linear complexity and leads to the maximized video quality for the admitted clients. It also satisfies the energy budget and access time constraints.

Index Terms — Energy Efficient Transmission, Modulation and Coding Scheme, Resource Sharing, Scheduling.

I. INTRODUCTION

With the significant progress in video coding technologies together with the rapid developments of network infrastructures as well as the exponential growth in storage capacity and computing power, an increasing number of video applications employed a variety of transmission and storage systems that have been widely used in our daily life. Among these applications, video signals could be transmitted over wired/wireless channels with variable bandwidth; they might be stored on media with different capacities, including a range from low-capacity memory sticks to high capacity DVDs; they also could be displayed on a variety of devices, ranging from mobile phones with small screens to high-end systems with high-definition displays.

In wireless networks, communication takes place over a time-varying and unreliable channel. Video transmission over unreliable networks has been an active field of research. Error resilience and error concealment have received considerable attention [1], [2]. A complimentary approach is to adapt the behavior of the video encoder to the conditions of the channel. For example, in [3], the objective is to minimize the expected distortion at the receiver subject to rate constraints derived from a stochastic model of the wireless channel and application delay constraints. In [4]–[7], the approach is to select the coding mode for each macro block (MB) taking into account the probability of packet loss in the channel and the error concealment technique used by the decoder in order to reduce the expected distortion at the receiver.

In traditional video systems, it is always assumed that the bandwidth required by a video client will be guaranteed. An encoder just needs to compress the input video signal at a bit rate that is less than and close to the predefined bit rate, and the decoder reconstructs the video using all the bits received from the channel. However, in modern video transmission over the Internet, it is almost impossible for the encoder to know the available bandwidth in advance. The video should be encoded over a bit rate range instead of a given bit rate [28]. The conventional non-scalable video coding cannot be used for this type of applications and this gives rise to the need to have a scalable video coding technology.

The research on scalable video coding has been an active area for about 20 years. Many early standards, e.g. MPEG-2 Video/H.262 and MPEG-4 Visual, have included tools to provide several important scalabilities. However, the scalable profiles of these standards have rarely been used. One reason is due to the characteristics of traditional video transmission systems in which scalabilities is not really necessary [27]. Another main cause for the situation is the fact that scalability always comes along with a significant loss in coding efficiency as well as a large increase in decoder complexity compared to the corresponding non-scalable profiles.

Scalable video coding involves generating a coded representation (bit-stream) that allows decoding of appropriate subsets to reconstruct complete pictures of resolution or quality commensurate with the proportion of the bit-stream decoded [9], [21]. The minimum bit-stream subset that can be decoded is called base layer. The remaining bits in the bitstream are called enhancement layer(s) and by decoding the enhancement layer(s) more details are obtained to get the video at higher resolution or quality as compared to base layer.

The desire for scalable video coding, which allows on-the-fly adaptation to certain application requirements such as display and processing capabilities of target devices, and varying transmission conditions [15], originates from the continuous evolution of receiving devices and the increasing usage of transmission systems that are characterized by a widely varying connection quality.

Video coding today is used in a wide range of applications ranging from multimedia messaging, video telephony and video conferencing over mobile TV [24], wireless and Internet video streaming, to standard and high-definition TV broadcasting. In particular, the Internet and wireless networks gain more and more importance for video applications.
A video bit stream is called scalable when parts of the stream can be removed in a way that the resulting sub-stream forms another valid bit stream for some target decoder, and the sub-stream represents the source content with a reconstruction quality that is less than that of the complete original bit stream but is high when considering the lower quantity of remaining data [20]. Bit streams that do not provide this property are referred to as single-layer bit streams. The usual modes of scalability are temporal, spatial, and quality scalability.

Spatial scalability and temporal scalability describe cases in which subsets of the bit stream represent the source content with a reduced picture size (spatial resolution) or frame rate (temporal resolution), respectively [14]. With quality scalability, the sub-stream provides the same spatio-temporal resolution as the complete bit stream, but with a lower fidelity – where fidelity is often informally referred to as signal-to-noise ratio (SNR).

Quality scalability is also commonly referred to as fidelity or SNR scalability. More rarely required scalability modes are Region-Of-Interest (ROI) and object-based scalability, in which the sub-streams typically represent spatially contiguous regions of the original picture area. The different types of scalability can also be combined, so that a multitude of representations with different spatio-temporal resolutions [8] and bit rates can be supported within a single scalable bit stream. Scalable video coding is being developed in response to the need of robust video delivery for heterogeneous clients.

SVC encodes videos into multiple layers which can be judiciously discarded to achieve graceful quality degradation. More importantly, SVC [22] can cater for multi resolution demands of heterogeneous clients via a single scalable bit stream, whereby the clients receive and decode different fractions of the video bit stream to obtain the desired resolutions and qualities.

On the other hand, the broadcast nature of wireless medium allows the simultaneous transmission of a video packet to multiple receivers [8]. By establishing multicast groups, an SVC bit stream can be streamed to multiple heterogeneous clients who demand different video resolutions. One can exploit the advantage of multicast architecture to deliver scalable videos to multiple clients concurrently.

The video sources for most streaming applications are typically precoded stored video sequences with relative high bit rates. However, the currently deployed wireless cellular systems are designed to only support voice and lower bit rate data. In order to support video streaming over such networks, the high rate video sources need to be adapted through a variety of schemes [10], such as scalable video stream extraction, transcoding and summarization before they can be accommodated by the wireless channel.

Different video content segments have different rate-distortion characteristics, e.g., some segment may be part of an action movie and requires a lot of bits to encode, while others maybe news anchors talking that require relatively less bits to encode [10]. In a wireless multi-access channel, the type of multi-user content diversity in content rate-distortion characteristics should be taken into consideration while optimizing the network resource.

The resource consumptions of video users are typically discrete, i.e., measured in frames instead of in bits. As a result, their utility functions are discrete as well, and typically do not have close form representations [16]. Therefore, most of previous work on resource allocation for elastic data traffic does not directly apply here, and a new optimization framework is needed.

Wireless resource allocation and scheduling approaches can be categorized into two classes: i) time-division multiplexed (TDM) systems, where a single user is transmitted to in each time-slot and ii) systems in which the transmitter can simultaneously transmit to multiple users in each time-slot [13]. These systems employ a combination of TDM and another multiplexing technique such as CDMA or OFDM.

Delay-sensitivity, high-data rate and unequal priority of scalable video traffic pose challenges to the wireless video transmission. The limited battery life span in each mobile terminal and the scarcity of wireless bandwidth further complicate the streaming of videos over mobile networks [23]. Efficient resource allocation and scheduling for streaming of video over the resource constrained network remains an open issue.

In this paper, the problem of multiuser resource allocation and scheduling for SVC multicast over wireless networks was investigated. Specifically, consider the portable wireless terminals with limited power supply [19]. The SVC multicast is supported with an amount of energy budget. In addition, the SVC multicast is also constrained by the limited channel access due to the competitions from other traffics. Under the resource constraints, the optimized multicast strategy such that the video quality at the client side is maximized was investigated.

The problem of multiuser resource allocation and packet scheduling has been thoroughly investigated [9]-[13]. However, they were mainly concerned on the efficient scheduling and bandwidth utilization, whereas the energy efficiency in the transmissions was not considered. As higher bandwidth results in higher energy consumptions, the resource allocation for energy-constrained networks attracts our attention.

Optimal burst scheduling algorithm for mobile TV broadcast networks [24] explains about the conservation of energy. The energy efficient multicasting of scalable video over WiMax networks [25] proposed an approximation algorithm to a substream selection problem which maximizes the video quality and minimizes the energy consumption for receivers, subject to bandwidth constraint.
Fig. 1. (a) Normalized transmission times, (b) Normalized transmission energies.

In [29] jointly optimized the source and channel coding rates, number of subcarriers, and the transmission power for scalable video transmission over the generalized multicarrier CDMA system. However, it was limited to the unicast link, and the energy efficiency in the transmission was not considered. In [30], proposed a cross-layer architecture for scalable video multicast over WLANs.

II. MULTICAST CHANNEL PROBING AND MODELING

Multicast characterizes one-to-many transmissions. Due to differences of geographical location, channel noise/interference, fading and path loss, different receivers experience different channel conditions. To ensure reliable packet transmissions to all admitted clients, the client with the worst channel condition is to be determined. The worst channel condition of each multicast group is the reference channel condition to which the resource allocation algorithm should refer.

A. Multicast Channel Probing

Auto rate selection for multicast (ARSM) [27] is limited to videos encoded in a base layer and a quality enhancement layer. For each multicast group, the router initiates multicast probe (MP) frame which contains the information of the video bit stream and its resolution to all clients. The router then starts a timer to wait for the replies of multicast response (MR) frame.

Upon receiving the MP frame, if a client does not subscribe to the multicast group, the MP frame is simply ignored. Otherwise, an admitted client estimates the signal-to-noise ratio (SNR) and path loss of its wireless channel. Based on the SNR, the client selects a random backoff time slot within a given backoff time slot interval [27]. A backoff process is initiated before it replies with an MR frame.

B. Multicast Channel Modeling

From the MR frames, the mesh router obtains the worst channel statistics of a multicast group. Minimizing the transmission power does not always lead to the energy efficiency. By lowering the transmission power, the channel SNR deteriorates and leads to a higher loss rate [29]. Even if a slower and more robust MCS is used, it would take longer time to transmit the packet.

Given the transmission energy as $E = P \cdot t$, a lower $P$ may sometime be offset by a longer duration $t$. In addition, a longer transmission time is usually not permitted due to the delay sensitive nature of video traffic and the limited channel access [26]. Besides, it results in a lower network throughput.

In the best effort video multicast service, consider the situations whereby the network resources are insufficient for streaming the video with the highest quality [17]. Consider the following circumstances:

- There is less competition for channel access time from background traffics, but the energy budget is limited.
- There are sufficient energy budgets but limited channel access due to competition from background traffics.
- Both channel access and energy budget are insufficient.

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<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
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<tbody>
<tr>
<td>Channel</td>
<td>Wireless Channel</td>
</tr>
<tr>
<td>Propagation</td>
<td>Two Ray Grounding</td>
</tr>
<tr>
<td>Antenna</td>
<td>Omni Directional Antenna</td>
</tr>
<tr>
<td>MAC Address</td>
<td>802.11</td>
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<tr>
<td>Network Interface</td>
<td>PHY/Wireless Phy</td>
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<td>Routing Protocol</td>
<td>DSR</td>
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<tr>
<td>Number of nodes</td>
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<tr>
<td>Initial Energy</td>
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<tr>
<td>Idle Power</td>
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</tr>
<tr>
<td>Receiver Power</td>
<td>1.5</td>
</tr>
<tr>
<td>Transmitter Power</td>
<td>2.0</td>
</tr>
</tbody>
</table>

C. Utility Metric and Packet Loss Modeling

Video quality and quality degradation during transmission are often quantified in the video’s peak signal-to-noise ratio (PSNR) [22]. However, PSNR only describes the quality difference for videos with identical spatial-temporal resolution, and is not applicable for multi-resolution videos.

In video transmissions, distortion due to the loss of a packet can be computed as the quality degradation between the received video and the ground truth. Although
exhaustive computations reflect most accurately importance of a packet, it is computationally expensive. As the SVC coding structure expands in 3 dimensions (spatial, temporal, and quality), there are many possible dependency combinations.

As only a limited number of layers can be multicasted in a resource-constrained network, which layer to be included in the combination wanted to be determined such that the video quality is maximized [19], [16]. To do so, first need to determine relative importance of a layer. Due to the layered coding architecture, quality impacts of all video packets from the same SVC layer are approximately equal is assumed. Then, by determining the quality impact of an SVC layer, the quality impact of all video packets belonging to the same SVC layer can be approximated.

Layered coding results in inter-layer dependencies. Depending on the video content, SVC encoder determines the best inter-layer prediction, and results in an order of inter-layer dependency [32]. In the single-hop multicast, timing is properly controlled by the mesh router while the propagation delay is negligible. Assume no packet loss due to delay or buffer overflow.

Table 2: Data Rates (KBPS) and PSNR Values (dB) of the Scalable Videos used in the Evaluation

<table>
<thead>
<tr>
<th>Name</th>
<th>1 Layer</th>
<th>2 Layer</th>
<th>3 Layer</th>
<th>4 Layer</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>r_1</td>
<td>q_1</td>
<td>r_2</td>
<td>q_2</td>
</tr>
<tr>
<td>FOREMAN</td>
<td>170</td>
<td>32.9</td>
<td>407</td>
<td>34.86</td>
</tr>
<tr>
<td>BUS</td>
<td>185</td>
<td>33.17</td>
<td>390</td>
<td>35.43</td>
</tr>
</tbody>
</table>

III. DRASM ALGORITHM AND Q1 SCHEDULING

Propose the Dynamic Resource Allocation Selection for Multicast (DRASM) algorithm to solve the resource allocation problem of SVC multicast. Using DRASM algorithm wasted energy is calculated for each and every node. For energy-efficiency, an efficient solution should minimize the waste of energy. The wasted energy $E_{\text{wasted}}$ is defined as the total energy used to transmit a lost/corrupted packet and its dependencies.

Using this algorithm, the layer having higher packet loss ratio will be eliminated. Transmission energy and transmission duration are the parameters of this algorithm. Although 0-1 Knapsack problem could be theoretically via techniques such as the branch and bound approach [24], it is too computationally intensive. The expected wasted energy is given by,

$$E [E_{\text{wasted}}] = (1-p_k) \sum_{i=r_k}^{K-1} E_i + p_k E [E_{k+1}^{\text{wasted}}]$$  \hspace{1cm} (1)

In (1), the expected wasted energy is computed as the probability for both-successful and unsuccessful transmission scenarios. Unsuccessful transmission leads to outright waste of energy for all dependent layers. Successful transmission is equivalent to the expected wasted energy of the next enhancement layer.

IV. SIMULATION RESULTS

A. Simulation Setup

Simulate the video multicast within a mesh cell using NS-2. Consider 30 mesh clients surrounding it. The videos are encoded into H.264/SVC bit streams using the SVC codec JSVM 9.8. Thirty clients are randomly distributed within the service cell to subscribe to two videos of two resolutions.

In existing method, DRASM method is compared with HARM. In HARM, the transmission power was fixed while the MCS was selected according to the ARSM protocol. To increase the throughput, HARM selected a higher MCS for enhancement layers, and expected the clients with a good channel SNR to gain a better video quality [20]. HARM assumes a fixed transmission power.

B. Simulation Results

In this section, the performance of Server Independent System (SIS) and Server Dependent System (SDS) is evaluated using Network Simulator. The performance of SIS is compared with the performance of SDS under various parameters. This analysis can be obtained by varying the parameters such as throughput, energy consumption, packet delay and packet delivery ratio with the simulation time.

The results are demonstrated using 30 nodes. Three types of resolution nodes are used such as Low Resolution Nodes (LRN), Medium Resolution Nodes (MRN) and High Resolution Nodes (HRN).

Fig. 2 represents packet delay performance of SIS and SDS. From Fig. 2, delay decreases by increasing simulation time in SDS. The delay of SDS decreases because the nodes receive data from the server. But in SIS, the nodes have to communicate with the nodes of its same resolution. Then sharing and broadcasting takes place. Due to this delay performance increases in SIS.

Fig. 3 represents throughput performance. Using SIS, the throughput decreases when compared with SDS (i.e.) all the nodes share its data with the server. The receiver node can directly request the server and then receive the data from the server. It will automatically increases the number of bits received per unit time.
Fig. 2: Packet Delay Performance

Fig. 3: Throughput Performance

Fig. 4: Packet Delivery Ratio Performance

Fig. 5, Fig. 6, Fig. 7 represent the Energy saving performance of different resolution nodes of both SIS and SDS. In SIS method, the energy saving will be more. From figures, the energy saving will be high in SDS because nodes directly receive data from the server. For example, STL nodes save more energy when compared with MTL and HTL resolution nodes. Because STL nodes receive data directly from the server but MTL and HTL resolution nodes receive data from the nearby neighbor nodes of same resolution nodes. It will increase the performance.
Similarly STH nodes save more energy when compared with MTH and LTH resolution nodes. Because STH nodes receive data directly from the server but MTH and LTH resolution nodes receive data from the nearby neighbor nodes of same resolution nodes. Likewise STM nodes save more energy when compared with LTM and HTM resolution nodes. Because STM nodes receive data directly from the server but LTM and HTM resolution nodes receive data from the nearby neighbor nodes of the same resolution nodes.

The data present in SDS is broadcasted, communicated and shared their information using a server. It will increase the energy saving and decrease the access time. Complexity will be less as compared to other video multicasting. The applications of video multicasting are video streaming, distance learning, video conference, video on demand, live video, IP surveillance systems and interactive gaming.

The work in this paper can be extended in different directions. For example, currently two bit streams have been taken into account. There are many bit streams are available such as Crew, Football, Mobile, City, Harbour, News, Soccer, ice data rates can be used and can analysis the parameters such as packet delivery ratio, packet delay, throughput, energy saving and packet drop.

ACKNOWLEDGMENT

The authors would like to thank the anonymous referees for their constructive comments and valuable suggestions which would help to improve the quality and presentation of the paper.

REFERENCES


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