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NETWORK ADAPTIVE LAYERED MULTICAST FOR HETEROGENEOUS WIRELESS AD HOC NETWORKS

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Layered transmission is a promising solution to video multicast over Heterogeneous Wireless Networks. However, since the number of layers is practically limited, noticeable mismatches would occur between the coarse-grained layer subscription levels and the heterogeneous and dynamic rate requirements from the receivers. In this paper, we propose a Network-adaptive Layered Multicast (NALM) approach, that exploits the increasing computing and communications capabilities of wireless devices. We show that by having few multicast nodes (about 10% the number of receivers) to encode and decode the video, an improvement of more than 30% in bandwidth efficiency could be achieved. Furthermore, due to the proximity of such encoding/decoding nodes to the receivers than the source itself, more accurate and faster evaluation of network conditions would be possible leading to faster convergence and further improvement in efficiency.

Keywords: Adaptive Multicast, Wireless networks

1 Introduction

The future Internet will offer ubiquitous wireless connectivity. Wireless adaptive networks will proliferate at the edges of the Internet and will enable novel applications [1]. Wireless devices will include laptops, PDAs, and in particular cell phones (over 2 billion in use).

An interesting scenario of future wireless Internet, is that of multiple radio devices, which are in close physical proximity and can collaborate by forming ad-hoc networks [2, 3]. For example, wireless devices, such as cell phones, PDAs, etc., in various environments can set-up an ad-hoc network to improve coverage and the communication quality. Another example, in an emergency situation, in which the cellular infrastructure has failed, cellular phones can form an ad-hoc network between themselves to provide vital information to people. Another application is that of communication among cars on the highway, cluster of cars can communicate in an ad-hoc mode to share information that is delivered in specific points of the highway.

An important part of the information that will travel over the future wireless ad-hoc Internet will be video multicasting, which will span from short video fragments, interactive maps up to full video on demand.

The major problem in delivering video over wireless ad-hoc networks is the high bandwidth variation, which depends on dynamic changes of wireless link quality and network topology. Therefore, the solutions for video multicasting over wireless ad-hoc networks should be able to adapt to the changing network conditions in order to achieve the optimal tradeoff between video quality and resources being used in networking and computing.

In designing such adaptive networking protocols for multicast video over wireless ad-hoc networks, an important factor is the increasing computing and communications capabilities of wireless devices, such as cell phones, PDAs, etc. Therefore, by using decoding/encoding (transcoding) capabilities in wireless devices, it is possible to design protocols that pragmatically take advantage of the momentary link and network conditions by generating adaptive video streams. However, scalability and energy limitations are important constraints to be accounted for. So, the level of knowing link and network conditions will depend on the amount of feedback, which is overhead to the user data. On the other hand, most of the involved wireless devices will be running on batteries, which will limit their capacity of decoding/encoding. Another factor to be taken into account is the heterogeneity of the network, which will include a wide range of devices with different capabilities. Therefore, adaptive protocols for video over wireless ad-hoc networks should offer to users the possibility to select the desired tradeoff among quality and constrains on scalability and energy consumption. Combining the concepts of layered video and transcoding, the needs of wireless networks could be met more efficiently.

In general, the wider the bandwidth range that needs to be covered by a scalable video stream, the lower the overall video quality is [4]. With the aforementioned increase in heterogeneity over emerging wireless networks, there is a need for scalable video coding and distribution solutions that maintain good video quality, simultaneously addressing the high level of anticipated bandwidth variation over these networks.

We propose a network adaptive protocol, called Network Adaptive Layered Multicast for Heterogeneous Wireless Networks - NALM. NALM relies on the capability of wireless nodes to act as transcoders. Moreover, the nodes can adapt the video stream to the dynamic conditions of their part of multicast tree [5]. Therefore, NALM can be viewed as a generalization of some recent work on active based networks with (nonscalable) video transcoding capabilities of MPEG streams. NALM supports the argument for active services that we can best support or enhance many applications using information or intelligent services only available inside a network.

There are multifold advantages of NALM. NALM addresses heterogeneity and improves fairness/efficiency when compared to existing schemes by up to 35%. Also, with NALM, it is possible to infer network conditions like bandwidth, congestion, packet losses and latency more accurately than from the source itself. Also, the proposed NALM system falls under the umbrella of active networks. Thus, NALM can be viewed as a generalization of some recent work on active based networks with (nonscalable) video transcoding capabilities.

The remainder of the paper is organized as follows: Section II presents some related work. Section III presents some background related to receiver-driven layered multicast (RLM) [6] and on realizing to transcoding in real time. Section IV formulates the problem and Section V presents the protocol. In Section VI we evaluate the performance of our NALM and finally we conclude in Section VII.

2 Related Work

QoS multicast routing has been an active research area for many years. Most of these problems belong to the class of minimum or constrained minimum Steiner tree problems [7], which are well-known to be NP-complete. We refer readers to [8] for a comparison study and [9] for a survey of multicast routing protocols in ad hoc networks.

Layered representations for Internet streaming have been widely studied, e.g., in [6, 10, 11]. In addition, scalable representations have become part of established video coding standards, such as MPEG and H.263+ [12, 13]. Recent advances in layered coding [14, 15, 16] have demonstrated that fine-tuning the layer rates can be efficiently implemented with fast response times and low overhead.

Much attention has also been paid towards extending scalable video streaming for multicast applications in wireless networks [17, 18, 19, 20]. We direct the reader to [21] for a very nice survey on several approaches for multicast over wireless mobile ad hoc networks. Liu et al. provide a comprehensive summary of the mechanisms used in video multicast for quality and fairness adaptation, as well as network and coding requirements [22].

Multiple description coding has been proposed as an alternative to layered coding for streaming over unreliable channels [23, 24]. Each description alone can guarantee a basic level of reconstruction quality of the source, and every additional description can further improve that quality. However, in [25, 26], it has been shown that layered coding with good allocation outperforms multiple description Coding over Multiple Paths. Apart from that, computing and maintaining multiple disjoint multicast trees imposes more overhead than multicasting through a single tree.

The most similar related work with ours would be [4] which proposes an approach based on Transcaling for addressing the bandwidth variation issue over emerging wireless and mobile multimedia IP networks. Transcaling represents a generalization of video transcoding. The authors proposed to have nodes such as gateways for wireless networks to perform Transcaling. The authors did not assume control over the placement and number of such Transcalers, which makes our work different and novel. To the best of our knowledge, we are not aware of any works that address the issue of Transcoding for wireless networks and study the efficient placement and number of transcoders in the network.

3 Background

A simple case of our proposed approach can be described within the context of receiver-driven layered multicast (RLM) [6]. Therefore, we first briefly outline some of the basic characteristics of the RLM framework, in order to highlight how this framework can be extended to NALM. Later in the section, we also present the recent advances in video compression technology that realize transcoding in real time.

3.1 Receiver-driven layered multicast (RLM)

McCanne et al. [6] proposed the first practical adaptation protocol for layered video multicast over the best-effort Internet. This protocol, known as receiver-driven layered multicast (RLM), is a pure end-to-end adaptation protocol and requires a FIFO drop-tail router only. A RLM sender transmits each video layer over a separate multicast group. RLM predetermines the number of layers as well as their rates. RLM performs adaptation only at the receivers' end by a probing-based scheme, where a receiver periodically joins a higher layer's group to explore the available bandwidth. If packet loss exceeds some threshold after the join experiment- that is, when congestion occurs-the receiver should leave the group. Otherwise it will stay at the new subscription level.

Similar to RLM, NALM is driven by the receivers' available bandwidth and their corresponding requests for viewing scalable video content. However, there is a fundamental difference between the proposed NALM framework and traditional RLM. Under NALM, some intermediate routers derive new scalable streams from the original stream. A derived scalable stream could have layers with rates different from the original scalable stream. The objective of the transcoding process is to improve the overall video quality by taking advantage of reduced uncertainties in the bandwidth variation at the intermediate nodes of the multicast tree.

3.2 Real-time Transcoding

A transcoder converts an existing video stream into a new stream with a different format or rate. A straightforward approach for transcoding is to decompress the video stream, process it, and then recompress it. This strategy is called spatial domain processing because it's performed on the original pixels. We can apply a rich set of operations in this domain-for example, pixel downsampling and color reduction. However, the efficiency of spatial domain processing is relatively low because it involves computationally intensive procedures for fully decoding and encoding.

Faster transcoding can be achieved by directly manipulating the compressed data in the frequency domain. This is feasible since state-of-the-art compression standards have similar processes. Examples include frequency filtering and quantization scale adjustment. For resolution downsampling, the motion vectors in the original stream by interpolation can be reused.

For instance, [27] presents the design of a video codec incorporating wavelet altering for spatial layering and repeated quantization for SNR layering. CPU-usage measurements on a Sun Ultra-1 workstation with one 167MHz UltraSPARC CPU and 128MB RAM were presented. A user with requirement for 96*80 pixels would have a bandwidth requirement of about 5kbits per frame, corresponding to 25 fps on a 128 kbps ISDN connection. On average, the encoder uses 25.4ms to construct and compress all 21 layers, corresponding to 13.7Mbps. This enables the encoder to process 39 frames per second, which is more than fast enough for real-time software encoding. Similarly, the total average decoding time is 34.81ms or 28fps. Further, in most real-life applications especially related to wireless networks, one would rarely reconstruct all 21 layers, meaning even faster decoding times.

Recent advances have enabled in much faster coding/decoding techniques [28, 29]. Studies have shown that, for a small number of streams, transcoding doesn't significantly increase the end-to-end delay because we can perform it within the interframe display time [29]. Taking all above factors into account, we believe that transcoding can be an effective and practical complement to receiver-driven adaptation.

4 Problem Formulation

Let $\bar{l}$ denote the *bandwidth allocation vector* for the layers, $\bar{l} = (l_1, l_2, \ldots, l_n)$, where n is the number of layers and l_i is the bandwidth of layer i. Let denote $\overline{L}$ the *cumulative bandwidth allocation vector*, $\overline{L} = (L_1, L_2, \ldots, L_n)$, where $L_i = \Sigma_{j=1}^{j=i} l_j$.

For a given $\overline{L}$, a receiver r with bandwidth b_r subscribes to the best matching cumulative bandwidth given by:

$$\phi = max_{l_i < b_r, l_i \in \overline{L}} l_i \tag{1}$$

Quantitative evaluations of protocols have relied on the normalized bandwidth, or fairness index. The fairness function of an individual receiver normalizes its actual received video rate by its expected bandwidth, and the global index of a session is the weighted average of the individual measures of all the receivers.

We measure the fairness for a receiver r with bandwidth b_r receiving a video stream of bandwidth v as:

$$F(b_r, v) = \begin{cases} \frac{v}{b_r}, & 0 \leq v \leq b_r \\ 0, & v = 0 \text{ or } v > b_r \end{cases} \tag{2}$$

A fairness index of 1 is optimal since it allows the receiver to fully exploit its available bandwidth. For a receiver with bandwidth lower than base layer video, the fairness is zero as it cannot receive any video at all. Others are between 0 and 1, and non-decreasing with respect to b_r.

Now, the problem would be to select $\bar{l}$ so as to minimize the *Global Fairness Index (GFI)* as the average fairness of all receivers. Assume the source knows the end-to-end bandwidths of all receivers. In such scenario we make following observation:

Observation: In the optimal layer bandwidth allocation, $L_i = b_t$, where $L_i \in \overline{L}$ and b_t is bandwidth of some receiver t.

Proof: Consider a scenario where some cumulative layer bandwidth L_i is not equal to any receiver's bandwidth. Without loss of generality, we assume $b_1 < b_2 < \ldots < b_m$, where m is the number of receivers. Let $b_{t-1} < L_i < b_t$. Now, increasing L_i to b_t would only result in increase in the *fairness* of each receiver with bandwidth b_q ($q \geq t$).

Hence, the optimization problem could be seen as a combinatorial problem of finding the set of $b_i^{'}$ for $(1 \leq i \leq n)$, such that $b_{i-1}^{'}$ and $b_i^{'}$ is bandwidth of some receiver. This problem could be solved by a dynamic programming algorithm with time complexity $O(nm^2)$.

However, it should be noted that when $m > n$, *Global Fairness Index* would be smaller than one. We also observe that, in practice, there is a lower bound for the rate of each layer l_i. Many compression tools simply specify a fixed rate for the base layer. In general, once the number of layers is fixed, the higher the number of receivers, the lower the *Global Fairness Index* would be.

By introducing *Transcoders* in the network, the problem could be restated as follows: How to simultaneously minimize the number of *Transcoders* while maximizing the *Global Fairness Index* across all the receivers.

5 Network Adaptive Protocol

In this section, we first present an example to illustrate how *Transcoding* could improve the efficiency. Then, we proceed to present our approach for selecting transcoders in the network to improve overall *GFI*.

5.1 Motivation

Consider a hypothetical but simple multicast session as presented in Figure 1 with four receivers and two layers with bandwidths l_1 and l_2.

Case 1 - *No transcoding*: In this case, source S encodes the video into two layers with bandwidths 100kbps and 200kbps, resulting in average fairness of 0.1875.

Case 2 - *With transcoding*: In this scenario, nodes N_1 and N_2 act as transcoders. Now, the source S encodes the video into two layers with bandwidths 200kbps and 200kbps, respectively. N_1 subscribes only to the base layer l_1, splits this video further into two layers l_1^1 and l_1^2. R_1 subscribes to l_1^1 and R2 subscribes to both l_1^1 and l_1^2. N_2 subscribes to both l_1 and l_2, transcodes them into and with rates 300kbps and 100kbps, respectively. R3 subscribes to and R4 subscribes to both and . Thus, transcoding could yield dramatic improvements in efficiency (fairness). In the above scenario transcoding yields a fairness of 1.

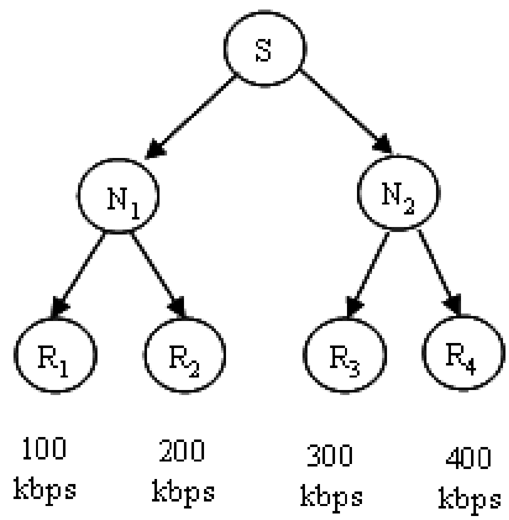

Fig. 1. A hypothetical layered multicast session with two layers and four subscribers.

5.2 Network Adaptive Layered Multicast (NALM)

In this section, we present NALM. NALM mainly involves three tasks: (a) selection of *transcoders*, (b) the process of *transcoding* itself and (c) merging the feedback at each stage to alleviate the feed implosion problem. We elaborate each task.

5.2.1 Selection of Transcoders

The challenge is to select minimal number of multicast routers to be transcoders. There is definitely a trade off between improving efficiency and keeping the end-to-end delay low. For instance, selecting more transcoder would yield higher efficiency. At the same time, since transcoding requires non-negligible duration, this contributes to the end-to-end delay and hence number of transcoders needs to be minimized.

We propose that a node performs transcoding, if the average fairness of the receivers in its subtree is below some threshold Th. This criterion would then ensure that the total fairness is within the application's specified limits.

5.2.2 Feedback Merging

Each multicast node that receives multiple feedbacks (either directly from different receivers or from some downstream intermediate nodes) merges the feedbacks into a single feedback and forwards to its upstream multicast node.

A multicast node m computes a feedback with following fields:

- max b_r, $\forall r \in subtree(m)$, b_r is the bandwidth of receiver r that belongs to *subtree(m)*, the subtree of m.
- Total number of receivers in the subtree, n_m
- Average fairness among the receivers in the subtree, F_m

5.2.3 Transcoding

A multicast node m decides to transcode the video if $F_m > Th$. The node then decides the optimal allocation of layer rates based on the bandwidths of the receivers and other transcoders in its subtree. A receiver's bandwidth is considered only if it does not belong to the subtree of any other transcoder in *subtree(m)*.

5.2.4 Discussion

We note that by having *Th* set to zero, no transcoders would exist and NALM exactly behaves like the underlying layered multicast protocol, for instance RLM. Also, *Th* could be used by the application/network administrator to specify the required fairness. The *Threshold* could also be adjusted to vary the number of Transcoders in the network as we illustrate in Section 5.1.

The convergence time is a metric used to evaluate the stability and scalability of a multicast system [6]. Here, we observe that NALM achieves convergence faster than the underlying multicast session mainly due to two reasons: First, in NALM, each *Transcoder* (including the Source) would have to deal with much smaller number of receivers than with RLM (for instance) with out any transcoding. Second, the *Transcoders* would be able to adapt to network conditions faster and more accurate because of lesser RTTs (Round Trip Times) to the receivers.

6 Performance Evaluation

In this section, we examine the performance of the NALM protocol under a variety of configurations. For comparison, we also present the results for the non-adaptive case, in which there are no transcoders present. Initially, the layer bandwidths are uniformly allocated i.e., the rates of all enhancement layers are equal.

Our study targets large groups. We vary the number of receivers from 10 to 100. The bandwidth of each receiver is randomly assigned between 50KBps and 1000KBps. This range covers the bandwidths of many available network access and video compression techniques. It is also a typical dynamic range of existing layered coders, such as the MPEG-4 PGFS.

Our study mainly focuses on following: Effect of threshold *Th* on efficiency and number of *Transcoders* and performance of NALM in different networks with different number of receivers. For each scenario, simulation experiments are repeated until the 90% confidence intervals of all average results are within 10%.

The effect of threshold is depicted in Figs. 2 and 3. As seen from Fig. 2, by varying the threshold to different values (between 0 and 1), desired performance could be achieved. Thus, *Th* could be used to make performance guarantees. Fig. 3 presents the number of transcoders needed to achieve the desired performance. It should be observed that higher the number of layers, the lesser is the number of transcoders. This can be explained from the fact that with more layers, each transcoder could meet the requirements of more receivers than with lesser layers. But, more layers would make transcoding itself more expensive and hence, the number of layers has to be limited and possibly low (less than 5).

An interesting observation is that NALM with two layers performs better than layer 4 and layer 3. To explain this, we need to study 2 and 3 together. Lesser number layers results in the number of transcoders to be relatively much higher (9 out of 25 receivers) than the case with higher number of layers (3 out of 25). As the objective is to minimize the number of transcoders, we focus our discussion further on NALM with three and four layers.

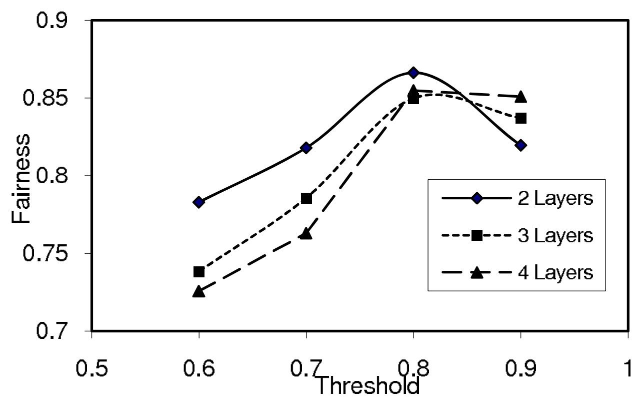

Fig. 2. Effect of Threshold Th on the fairness of NALM for different number of layers and 25 receivers.

Figs. 4, 5 and 6 present the performance results of NALM and non-adaptive scheme (RLM) in different networks with varied number of subscribers for different number of layers for both adaptive and non-adaptive scenarios. We note that with just around 10 to 15% of nodes being transcoders, there is an improvement of around 20 to 35% in fairness compared to non-adaptive scenario. We note that higher the number of layers, lesser the number of transcoders needed to ensure fairness guarantees as described previously in this section.

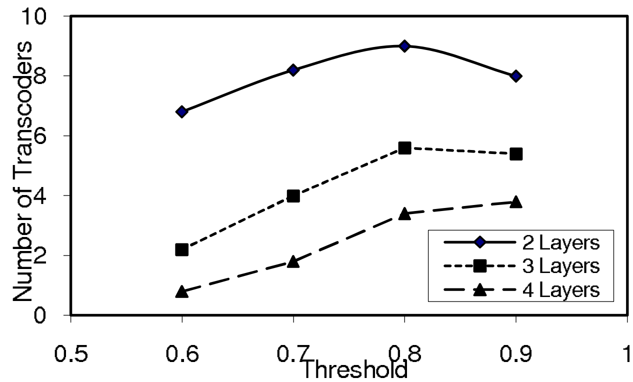

Fig. 3. Effect of Threshold Th on the Number of Transcoders of NALM for different number of layers and 25 receivers.

We also observed the maximum number of transcoders in the path to any given receiver. In scenario with 50 receivers, in most of the cases (about 70%), there was at most one transcoder, in general the maximum was two with four layered video. With three layers, the maximum was three in about 40% of scenarios and just two in the rest. In scenario with 25 receivers, we observe that maximum was just one node with four layers and just two with three layers. Thus, a conservative assumption that transcoding introduces a delay of 25msec still implies a maximum additional delay of 50msec to achieve about 35% improvement in fairness and ensure fairness gaurentees.

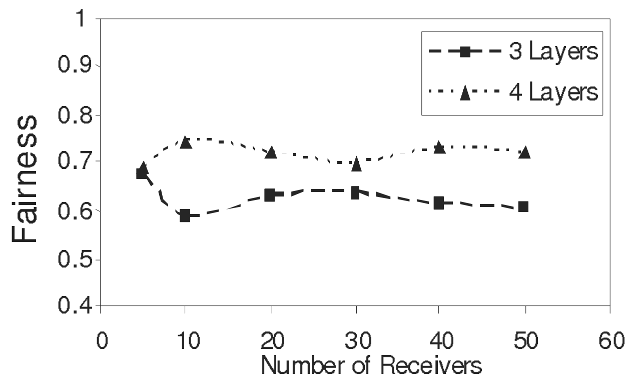

Fig. 4. Fairness of non-adaptive scheme for networks with three and four layers.

7 Conclusions

This paper presents NALM, Network-adaptive layered Multicast, to address the unique needs and challenges posed by heterogeneous wireless networks. We propose to have some of the network nodes to transcode the video before forwarding to other receivers. We reviewed the related advances in video compression techniques that enable this transcoding in near real-time. Performance evaluation shows that, by having very few nodes (about 10% the number

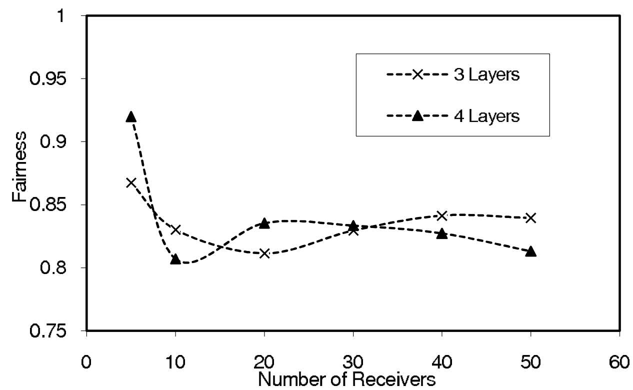

Fig. 5. Fairness of NALM scheme for networks with three and four layers.

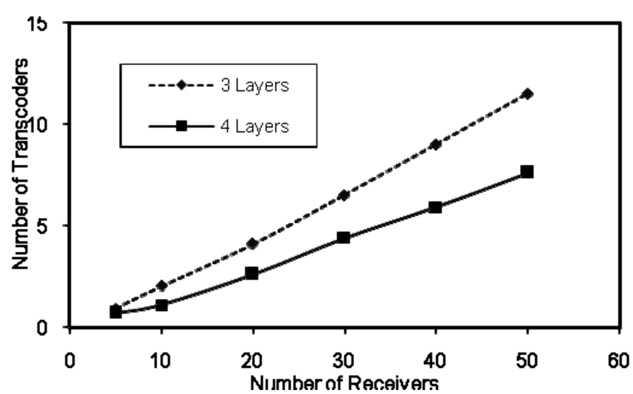

Fig. 6. Number of Transcoders of NALM scheme for networks with three and four layers.

of receivers) perform transcoding could yield an improvement in more than 30% in Global Fairness Index. There are also several other advantages like faster convergence and better evaluation of network conditions.

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