Network Coding for Advanced Video Streaming over Wireless Networks

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1. Introduction

During the last few years, thanks to the availability of low-cost high-capacity wireless connections, and with the increased computational power of mobile devices, the majority of services provided to mobile users has shifted from text-based to multimedia.

These days, mobile video services are proliferating at an astonishing pace: everything –from movies and TV shows to clips from ordinary users– is available to whoever is connected to the Internet, whether with a laptop, a tablet, or a smartphone. The new frontier of networking lies in this new paradigm: "Video Anywhere at Anytime".

Even though wireless technology has greatly advanced during the past years, a great deal of improvement is still needed in the domain of mobile video networking. Wireless networks still have a significantly lower capacity and a higher expected packet loss rate than wired networks, resulting in generally unreliable time- and location-varying channel conditions. Also, mobile terminals often rely on battery power, which is a scarce resource, and are far less dependable than Internet servers, routers, and clients.

This calls for video streaming techniques that on one hand reduce the bit-rate needed for transmission at a given video quality, and on the other hand are capable to provide a graceful degradation in presence of losses.

2. Network Coding

One of the fundamental assumptions of classical networking is that multi-hop data transfers are handled at intermediate nodes by forwarding the received messages without modifying their content. If more data flows share an intermediate node in their path, this will simply assign each of them a priority (scheduling) and an output link through which to be sent (routing). This view has been challenged with the introduction of the Network Coding (NC) paradigm [1,2], in which each message sent on a node's output link is a linear mixture, in a finite field, of the messages received on the node's input links. Such a strategy of packet mixing (or "coding"), together with means of decoding at the receiver, has been shown to outperform traditional routing by improving the throughput, minimizing the delivery delay, and reducing the impact of losses.

In this letter, we summarize some of our main contributions in the context of NC for high-quality video distribution services over wireless networks. In particular, we present our efforts of integrating NC with advanced video coding techniques such as Multiple Description Coding (MDC), which is used to provide a graceful degradation in the presence of losses in the stream, and Multi-View Coding (MVC), which is used to provide new and interactive 3D video services to the users. We also discuss how the overhead due to the use of NC can be reduced, thus better accommodating the relatively small MTU used in wireless networks.

3. Joint MDC/NC Streaming over Wireless Overlay

Multiple description coding is based on splitting a media stream into a certain number of sub-streams, known as descriptions. Any description can be independently decoded, but the quality increases with the number of descriptions and can be expected to be roughly proportional to the bit-rate sustained by the receiver. MDC is considered a valuable tool to cope with packet losses in wireless networks [3].

In our work [4], we proposed to use MDC jointly with NC to allow instant decoding of received video packets. We first formulated the problem of broadcasting a video stream encoded in multiple descriptions over a wireless network in terms of finding an optimal set of coding coefficients; then, we introduced an objective function that takes into account the effects on the total distortion of decoding a given number of descriptions.

The optimal encoding coefficients are selected via a distributed maximization of the objective function, which the nodes in the network operate based on up-to-date information about the topology. This information is gathered through a wireless overlay construction and maintenance cross-layer protocol we had previously proposed for real-time streaming of MDC video [5,6].

Our experimental results (Fig. 1) show that this approach consistently outperforms the well-known random linear network coding technique [7].

Arguably, this is due to the limits on the generation size imposed by the delay constraints that severely affect the performance of the reference technique.

4. Scheduling for Streaming MDC/MVC Content over Wireless Networks

While the method presented in the previous section benefits from a transmission overlay that supports the exchange of information among nodes, we present here another contribution wherein the optimization is performed without any feedback from the receivers. Namely, we propose a framework for video delivery in wireless networks that combines Expanding Window Network Coding [8], and a novel Rate-Distortion Optimized scheduling algorithm that optimizes the order in which the video packets are included in the coding window. We successfully applied this framework to both MDC [9] and Multi-View streams [10].

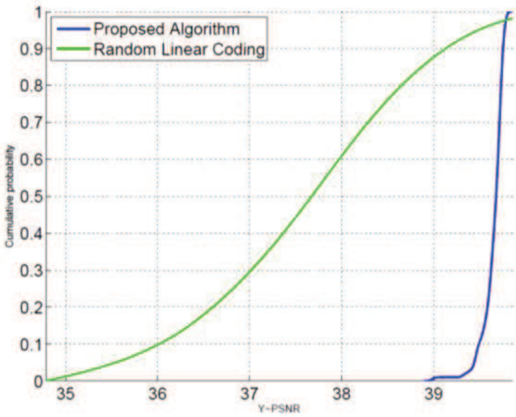

Figure 2 Comparison of PSNR cumulative distribution functions for video sequence "foreman" CIF, 30 fps, 1.8 Mbps

Expanding Window Network Coding (EWNC) is a NC strategy that progressively increases the size of the coding window by using a lower-triangular mixing matrix. The order of inclusion in the coding window is crucial as, by using Gaussian elimination at the receiver side, this method provides instant decodability of data packets.

Since the communication could be abruptly interrupted, due to the neighbors' mobility or disconnection, the scheduling has to be such that the expected video quality is maximized at each sending opportunity. However, imposing the optimal scheduling on all nodes would completely eliminate diversity, thus defeating the purpose of using NC. To address this challenge, we proposed to provide the nodes with a simplified RD model of the stream, so that parts of the video with similar RD properties are considered equivalent for the scheduling purpose (*clustering*). This provides them with a degree of freedom in the choice of the schedule, yet results on each node in a scheduling just slightly less performing than the optimal one.

Applied to both MDC and MVC streams, this strategy has shown to achieve a much higher video quality than both non-NC approaches, and NC approaches based on exact RD optimization or random scheduling (Fig.2).

5. Low-Overhead Network Coding

One of the commonly mentioned drawbacks of network coding is its high overhead. Since the decoder needs to invert the exact mixing matrix in order to be able to reconstruct the original packets, the senders have to include, in each mixed packet, the coefficient used in the combination.

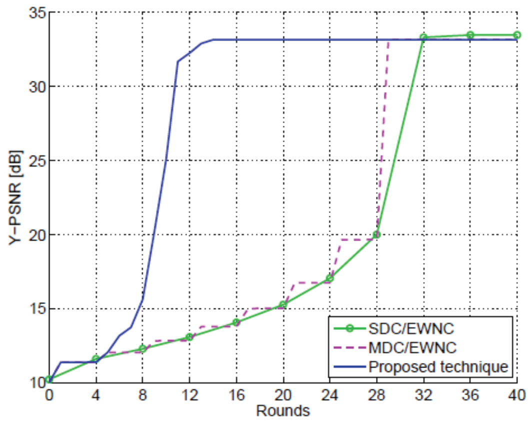

Figure 3 Comparison of average PSNR of the decoded sequences for two sources and 10% packet loss rate

In our recent work [11] we have argued that, using a combination of channel coding and a limited *a priori* knowledge of the sources, it is possible to reconstruct the original messages with high probability even if the combination coefficients are not sent.

This work is placed in the context of Blind Source Separation [12] –a well-established domain of research– but has to deal with the additional constraint that the sources are defined in a finite field, a very challenging addition that so far only few works have addressed [13].

Most BSS techniques rely on entropy minimization as a tool to distinguish between original sources (typically structured, thus carrying low entropy) and linear mixtures (less structured, and therefore with higher entropy). Unfortunately, in the case of video content, the encoded bit-stream has typically a distribution very close to uniform, *i.e.*, a very high entropy.

Our main idea is to increase the discriminating power of the algorithm by preprocessing the sources with an error-detecting code. The entropy minimization is then performed at the receiver, restraining the estimation of the entropy to the solutions that are admissible in the sense that the reconstructed source is a codeword. This eliminates several solutions that, even if they present low entropy and could be mistakenly identified as sources by merely entropy-based techniques, cannot be admitted as they are not part of the code. Ideally, the code should be such that only the original sources belong to the code, whereas any other possible mixtures do not. This is in practice unfeasible, but we design a code such that the probability of a mixture accidentally being a codeword is very low.

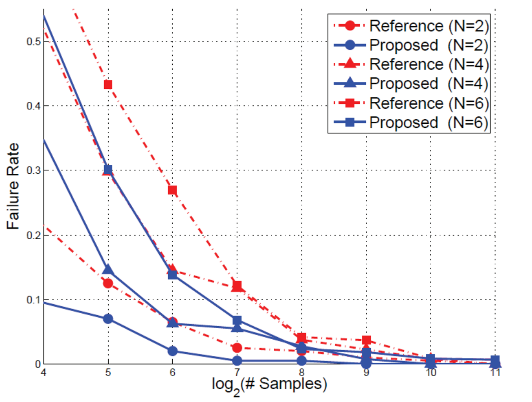

Figure 4 Comparison between the entropy-only BSS method and the proposed technique for finite field GF(4) as a function of the number of samples in the mixture

Our experimental results show that the proposed technique consistently outperforms the entropy-based method, especially in the case of sources with a small number of available samples, which is more critical for the entropy-based methods, making our BSS method more suitable for practical wireless applications, where the number of samples is typically limited by the size of a packet.

6. Conclusion

In this letter we have shown some applications of network coding to multimedia streaming on wireless networks. Our results confirm that NC has the potential for improving the video streaming services on wireless networks, by increasing the throughput and reducing the delay with a slight packet overhead.

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