

## Network Coding meets Multimedia: Opportunities and Challenges

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**Abstract:** *Network coding is a new paradigm that extends the traditional routing by allowing intermediate nodes in a network to combine packets, thus increasing the information per packet. Network coding research has –for the most part– considered flows from an information theoretical perspective, or data flows and content distribution scenarios. In this article, we focus on the issues that arise when network coding is applied specifically to multimedia streams. We identify some unique challenges and opportunities that arise in this case, and we give an overview of existing work as well as of open research questions in this area.*

### Network Coding

Today's networks are based on the fundamental principle that the network forwards data but the information itself is only processed at the end-systems. Network coding is a research field that emerged over the past decade and breaks this fundamental assumption: it advocates that, in addition to forwarding packets, intermediate nodes should be allowed to also process and recombine several incoming packets into one or more outgoing packets [1,2]. This breakthrough idea has spawned a significant effort [3,4], initially in the information theory and computer science communities and more recently in the networking and multimedia communities.

From a theoretical point of view, researchers have been studying the design of coding schemes and quantifying the benefits (in terms of throughput, delay and robustness) as well as the cost of network coding, for various traffic scenarios and network topologies. From a practical point of view, researchers are exploring potential applications to practical networking problems at various layers of the protocol stack. For example, in the context of wireless mesh networks, one-hop opportunistic network coding has been shown to increase throughput by mixing packets from different flows into a single packet and by broadcasting over the wireless medium [5,6]. In the context of peer-to-peer content distribution, random network coding has been shown to facilitate distributed scheduling, reduce the download times and increase robustness to node failures [7,8]. In the context

of TCP, window-based intra-flow network coding combined with retransmissions has been shown to successfully mask wireless losses from TCP congestion control [9]. There is also a growing body of work within the multimedia community that studies network coding techniques for multimedia and delay-sensitive traffic.

In this short article, we are interested in this last question: what are the novel challenges and opportunities that arise when network coding is applied to multimedia streams in particular? We review some of the existing work and identify some open research questions. *Disclaimer:* this is by no means a complete survey of the subject but is rather a view from our own research perspective.

### Network Coding for Multimedia

Below we discuss some fundamental properties of multimedia traffic and their implications for network coding.

**Unequal Packet Importance.** The fact that different packets, within the same media stream, have different contributions to distortion (due to video content, encoding, or playout deadlines) is well understood in the multimedia community. This fact lies at the heart of multimedia streaming: the unequal importance of packets is used to guide prioritized transmission over a network. Depending on the transmission scenario, available differentiation mechanisms are used to ensure that the most important packets of a particular stream are given priority, thus providing a graceful degradation in the presence of adverse network conditions. One challenge that arises from this fundamental property of multimedia, with respect to network coding, is that network coding, so far, has been agnostic to the content of the packets that are coded together.

In inter-session network coding, the goal is to mix together several packets from different flows, thus increasing the information per packet and eventually the throughput. However, for media streaming it is not only the quantity of delivered packets that matters but also their quality. In [10], we considered video traffic transmitted over wireless networks with opportunistic one-hop

network coding [6]. We designed video-aware network coding schemes that take into account both the decodability of network codes by several receivers and the unequal importance of video packets, namely, distortion values and playout deadlines. We demonstrated that these schemes improve the video quality without compromising the MAC throughput. In a sense, this work combined two orthogonal aspects of packet scheduling: (i) network coding to mix packets from different flows and increase throughput and (ii) radio-distortion optimized streaming of packets within the same stream to maximize video quality.

In intra-session network coding, the importance of packets coded together can be taken into account in the composition of generations, *i.e.*, in selecting which packets should be coded together [11]. In [11] Chou et al. also mentioned the idea of incorporating error protection in a generation, either in terms of simple redundancy or in more sophisticated ways, such as priority encoding transmission (PET), which is particularly well-suited for unequal error protection of layered media.

**Different Flow Characteristics and Requirements.** Moving from the granularity of packets to the granularity of flows, we observe that entire flows may also have different importance, *e.g.*, due to their traffic characteristics, sensitivity or pricing. When there are multiple media and/or data flows in a system, the question is which flows should be coded together? The rate and delay requirements of media streams should be taken into account when deciding which of them to code together and/or with data flows. For example, we might decide to not code audio traffic, due to its importance and tight delay requirement, but code together similar video streams. Some existing standards (*e.g.*, 802.11e or DiffServ) explicitly specify traffic priorities and allow for separation of media from data traffic, *e.g.*, via priority queuing or preferential dropping mechanisms. However, there is currently no explicit network coding differentiation mechanism. One approach would be to extend the framework in [10] and assign importance to packets based, not only on their distortion value and playout deadline, but also on their traffic type and priority.

**Delay Requirements.** Another inherent characteristic of media streaming and real-time communications is that they have strict delay requirements, which poses both a challenge and

an opportunity when network coding is used. On one hand, network coding increases delay due to additional encoding/decoding and possibly due to waiting at intermediate nodes for enough packets to arrive and be coded together. On the other hand, the increase in throughput can decrease the end-to-end delay. The design of scheduling and coding algorithms can trade-off throughput for delay so as to meet media requirements. In [10], we implicitly took the delay requirement into account, by incorporating it into the importance of a packet. In [12], Nguyen et al. looked at a downlink scenario and formulated the scheduling and coding problem within a markov decision process framework, which can also incorporate delay through its contribution to distortion. In the context of generation-based network coding [11], the throughput vs. delay tradeoff can be explicitly controlled by tuning the generation size.

**Rate Requirements.** One of the main advantages of network coding is that it extends the achievable rate region compared to traditional routing. Interestingly, this has a non-trivial interaction with the rate requirements and adaptation at higher layers. For example, video streams have their own, typically time-varying, rates that need to be adapted to match the rate region offered by the underlying network. Conversely, the rates at the video/application layer affect the availability of network coding opportunities at the underlying network coding layer and thus the achievable region. In our recent and ongoing work [13], we investigate video rate adaptation jointly optimized with network coding.

**Network Coding used for Error Correction.** Error resilience through redundancy and retransmissions is extensively studied in multimedia streaming. On one hand, network coding can be combined with well-known techniques, such as ARQ [9] and FEC [11,14]. On the other hand, network coding can be seen as an extension of FEC, applied not only at the source but also at intermediate nodes. In the context of peer-to-peer content storage and distribution, random network coding has been shown to be more robust than traditional FEC against failures or departures of nodes [7,8,15]. The intuition is that, in case of a block being lost, network coding produces unique innovative blocks, while FEC-based schemes can replicate the same block (original or redundant). Issues specific to peer-to-peer systems that use network coding and support video include: the need for

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low delay and continuous playout, the need for unequal error protection [16] and the interaction between video/FEC rates and the underlying rate region achieved by network coding.

### Summary and Directions

Media traffic has some characteristics and requirements (such as the difference importance of packets, the strict delay requirements and the time-varying video rate) that introduce unique challenges and opportunities for network coding. We advocate the need for cross-layer design of video-aware network coding schemes that specifically take these features into account. Combining techniques from network coding and media streaming can make the best of both worlds.

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