APPLICATION OF NETWORK CODING IN MULTIMEDIA COMMUNICATION

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ABSTRACT

Communication networks today share the same fundamental principle of operation. Whether it is packets over the Internet, or signals in a phone network, information is transported in the same way as cars share a highway or fluids share pipes. Network coding is an interesting paradigm that requests the network nodes to perform basic processing operations on packets in order to improve the throughput or the robustness of communication systems with network diversity. Itis a recent field in information theory that shows that instead of simply forwarding data, nodes may recombine several input packets into one or several output packets. This research has shown that the theory of network coding finds application in media streaming. Multimedia communication could typically benefit from network coding in overlay networks or wireless mesh networks. Coding decisions could also be adapted to the packet importance or the state of the receivers in order to maximize the quality of service.

Keywords: Multimedia communication, Quality of service, Network coding, Wireless mesh networks

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1.0. INTRODUCTION

A well-designed transmission model called network coding was proposed by Ahlswede et al.[1]. The key idea is that, one can approach the broadcast capacity of the network by allowing intermediate nodes inside the network to code and decode the information carried by the different flows. Since then, network coding's popularity is increasing and many research papers have appeared on the subject. In a traditional store-and forward network, packets are forwarded hop-by-hop along the intermediate nodes (e.g. routers) from a source to a destination.

An intermediate node forwards the packets as it receives through a predetermined path. On the other hand, network coding techniques allow an intermediate node to combine data from different input links before sending the combined data on its output links. This is called network encoding. For many problems such as multicast and broadcast, using appropriate encoding schemes at each intermediate nodes (typically linear combination of input data) can achieve the network capacity. Network coding technique can also be applied to wireless networks [2][3].The idea is to allow the intermediate network nodes to combine data received over different incoming links. Nodes with coding capabilities are referred to as encoding nodes, in contrast to forwarding nodes that can only forward and duplicate incoming packets. The network coding approach extends traditional routing schemes, which include only forwarding nodes.

In network coding, the network nodes combine the packets they receive before forwarding them to the neighbouring nodes. Intensive research efforts have demonstrated that such a processing in the network nodes can provide advantages in terms of throughput or robustness.These potentials, combined with the advent of ad hoc and wireless delivery architectures have triggered the interest of research community about the application of the network coding principles to media streaming applications.

This paper describes the potentials of network coding in Multimedia communication. A practical system based on network coding for file distribution to a large number of cooperative users was discussed. My approach does not require knowledge of the underlying network topology and, in addition nodes make decisions of how to propagate packets based on local information only.

By using network coding, nodes are able to make progress and finish a download even if the server leaves shortly after uploading only one copy of the file to the system and nodes depart immediately after they finish their download. Without network coding, if both the server and some peers suddenly depart the system, some blocks of the original file or of the source-encoded file will disappear and the remaining nodes will not be able to finish their downloads. This demonstrates that with network coding nodes are able to finish their download even in extreme circumstances.

2.0 MULTIMEDIA COMMUNICATION

Multimedia communication deals with the transfer, the protocols, services and mechanisms of discrete media data (such as text and graphics) *and* continuous media data (like audio and video) in/over digital networks.

Such a communication requires all involved components to be capable of handling a well-defined quality of service. The most important quality of service parameters are used to request (1) the required capacities of the involved resources, (2) compliance to end-to-end delay and jitter as timing restrictions, and (3) restriction of the loss characteristics.

Multimedia systems have attracted much attention during the last few years in the society as a whole and in the information technology field in particular. Multimedia communication comprises the techniques needed for distributed multimedia systems. To enable the access to information such as audio and video data, techniques must be developed which allow for the handling of audiovisual information in computer and communication systems.

There are 5 types of communication network that are used to provide multimedia communication services:

· Telephone networks

· Data networks

· Broadcast television networks

· Integrated services digital networks (ISDN)

· Broadband multi-service networks

Below shows different multimedia communication networks and their services

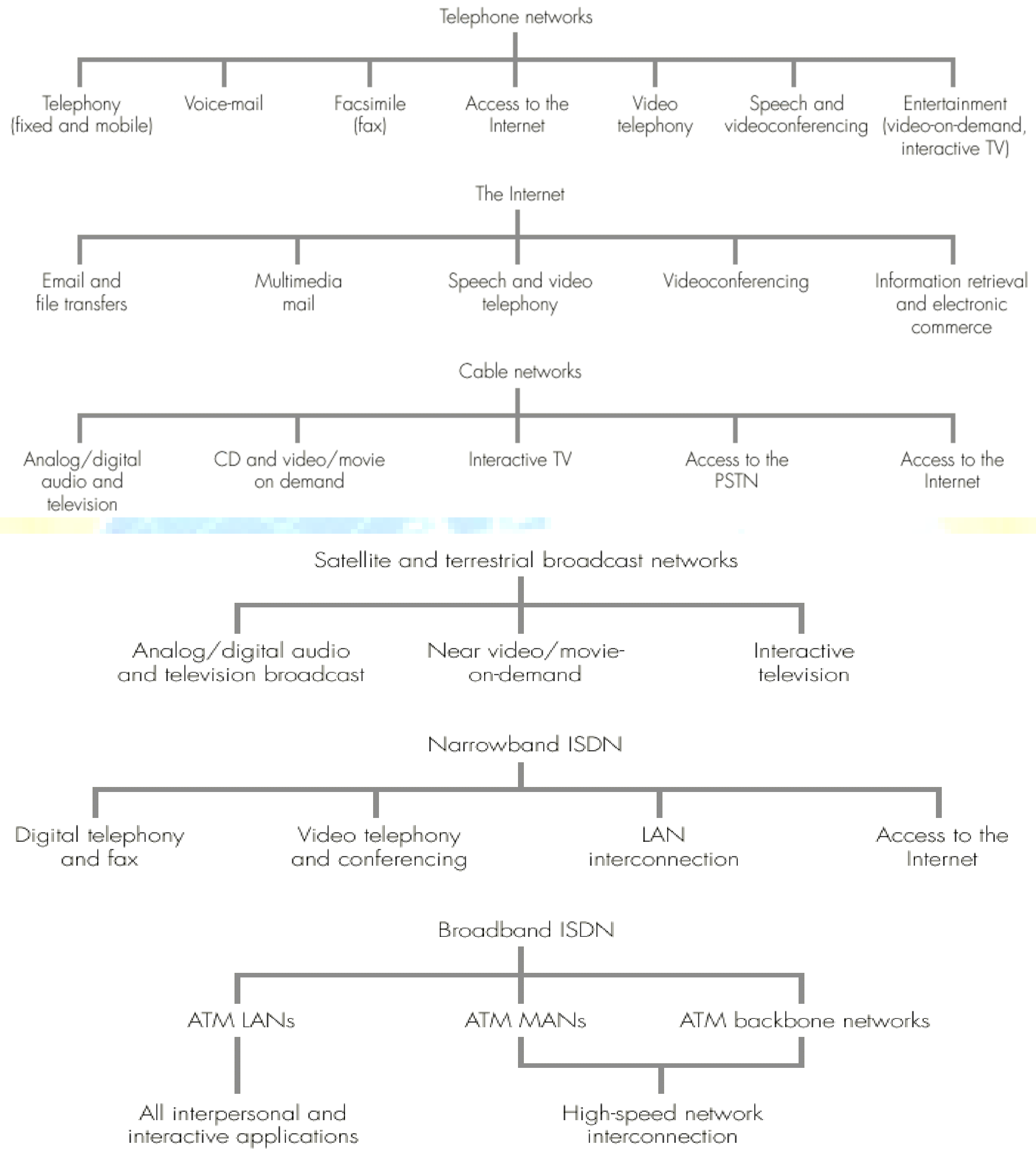

Fig.1: Different multimedia communication networks and their services

3.0. APPLICATION OF NETWORK CODING TO MEDIA STREAMING

The application of network coding algorithms to media streaming has to properly consider the specificities of multimedia communication applications, such as strict delay constraints, high bandwidth requirements, as well as the unequal importance of the data that further presents some tolerance to packet losses. The design of the system has to take all these parameters into consideration in order to produce efficient practical solutions with reasonable complexity.

When properly designed, network coding is able to take advantage of the network diversity that consists in multiple source peers and multiple transmission paths.

Network coding can be used to improve the throughput of a streaming system, to reduce the end-to-end delay, or to increase the robustness to packet loss, for example.It also provides an efficient solution that reduces the control overhead and avoids the need for reconciliation in distributed systems.

A. Peer-to-peer streaming

Network coding finds a perfect application in peer-to- peer multicast systems. Such systems have become recently very popular, as they rely on the bandwidth contributions from peers in order to reduce the load on the main streaming server. Multicast streaming is implemented by forwarding the media packets from the servers to all the clients via other peers that are grouped in an overlay or ad hoc configuration (see Fig. 2). The packet distribution is mostly organized in two modes, which are the push or pull strategies. In the first case, the packets are simply pushed through the different peers in a way that is determined by the senders. In the pull scenario, the clients request specific packets or group of packets from the source peers. Network coding can be beneficial in both cases, as it helps to cope with the network dynamics.

Multimedia streaming systems employing network coding techniques have been motivated by the success of Avalanche [4][5] which have been proposed for large scale content distribution in peer-to-peer networks. It envisions the deployment of huge overlay networks that allow fast downloading. This distributed architecture improves significantly the file download time of Bit Torrent [6].

One of the first works that has studied the performance of network coding in peer-to-peer (p2p) streaming has been proposed in [6]. Randomized linear network coding is implemented in a system called "Lava" in order to evaluate the tradeoffs and benefits of network coding in live p2p streaming. The system offers network coding as an option in a pull-based p2p streaming solution that allows for multiple TCP connections for multiple upstream peers. Prior to transmission, the streams are divided into segments of specific duration, similar to the idea of generation proposed in [7]. These segments are further divided into blocks that undergo network coding operations in the different peers. The peers periodically exchange messages to announce the availability of segments in a pull-based manner. At any time, peers make concurrent requests for segments that are missing in their playback buffer by addressing randomly one of the peers that possess the segment of interest. The peers then decode the segments from their playback buffer in a progressive manner using Gauss-Jordan elimination. The evaluation shows that the network coding scheme is resilient to network dynamics, maintains stable buffering levels and reduces the number of playback freezes. Network coding is shown to be most instrumental when the bandwidth supply barely meets the streaming demand.

Based on the encouraging results of [8], the same authors redesign the peer-to-peer streaming algorithm and propose the *R*2 architecture in [9]. In *R*2, randomized linear network coding is combined with a randomized push algorithm to take full advantage of coding operations at peer nodes. The peers periodically exchange buffer maps that indicate the segments that have not been fully downloaded yet. The *R*2 system sends the buffer maps together with the data packets whenever it is possible, otherwise they are transmitted separately. The frequency of these information's exchanges has to be chosen high enough, in order to avoid the transmission of redundant segments.

Whenever a coding opportunity is detected, a peer randomly chooses a video segment that the downstream peer has not completely received and generates a network coded block. The segment selection is inspired from [10]. The system also uses large segment sizes in order to avoid the transmission of too much overhead information by buffer map exchanges. The streams are progressively decoded by Gauss-Jordan elimination, similarly to the Lava system described above [11]. The *R*2 system provides several advantages in terms of buffer level and delay, as well as resilience to network dynamics.

The scalability of the system is also increased. Most of these advantages are due to the combination of push-based methods with randomized linear network coding.

The organization of the peers in the overlay network has a large influence on the performance of the streaming system. In particular, the delivery has to be organized in such a way that the bandwidth constraints can be respected, and such that the clients with the smallest bandwidth do not penalize the performance of the overall system. A method for constructing peer-to-peer overlay networks for data broadcasting is proposed in [12]. The overlay construction imposes that all the peers have the same number of parent nodes, which are the nodes that send them the data packets. Such a constraint tends to distribute the load over the network. Network coding is then used in the peer nodes for increased throughput and improved system robustness. One could also organize the overlay into several layered meshes. Heterogeneous receivers can then subscribe to some of the meshes depending on their capacities. The data is similarly organized into layers, and network coding is performed on packets of the same layers. The practical network coding scheme of [13] is adopted in this work due to its low complexity. The construction of the layered meshes takes into consideration the overlapping paths in order to exploit the network coding benefits. Depending on the network state and the clients' requirements, every receiver determines the proper number of meshes that it has to subscribe to. The network throughput is finally increased by network coding combined with appropriate mesh organization. In the same spirit, the work in [14] proposes to split the bit stream into several sub bit streams for streaming over peer-to-peer networks. A neighborhood management algorithm is then used to schedule appropriately the transmission of the different encoded sub-bit streams. Finally, the problem posed by the heterogeneity of the receivers could also be solved by combining network coding with multiple descriptionscoding as proposed in [15].

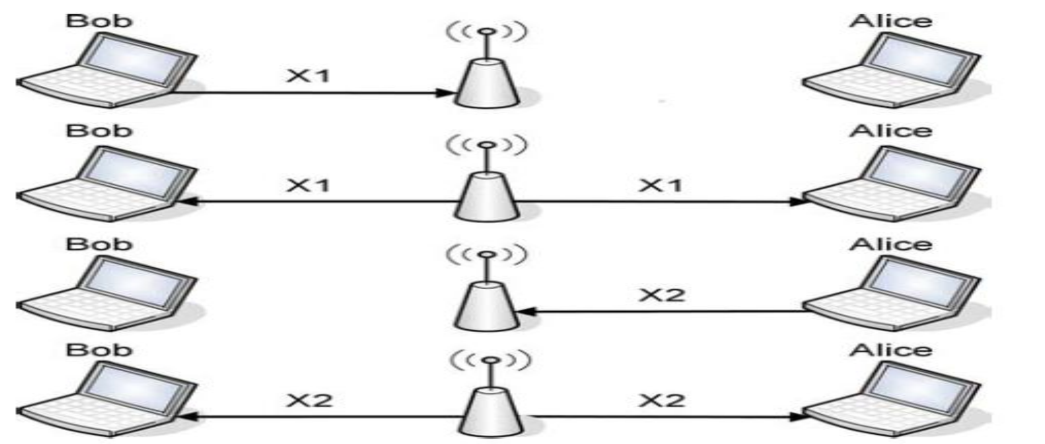

Fig.2. Wireless point-to-point communication.

B. Resiliency to packet losses

Network coding principles can also be used to increase the robustness of the streaming system. Multimedia streaming imposes in general strict timing constraints, which may render some of the packets useless if they are late at the receiver. Overall, the system has to be robust to packet erasures and maintain low delay for improved performance.

The increase of network throughput described in the previous section is only beneficial if the clients can decode the media packets.

One of the first attempts to realize some type of coding in the nodes of an overlay network is presented in [16]. The nodes are organized in multicast trees.

Some of them implement channel coding operations to increase the robustness of the system. These are called network-embedded FEC (NEF) nodes and perform Reed- Solomon (RS) decoding on the packets they receive. The decoded packets are encoded again with RS codes before transmission to the children peers. NEF nodes permit to increase the resiliency of the system, while avoiding a waste of resources with strong end-to-end protection. A greedy algorithm determines the number of NEF nodes and their location. Only a few well-positioned NEF nodes are sufficient to provide significant network throughput gains that result into a high video quality.

Similarly, decoding and encoding based on fountain codes is performed in the network nodes. The LT codes are used in this work since they perform close to perfect codes and eliminate the need for reconciliation among network peers and for packet scheduling. The intermediate network nodes wait for receiving a sufficient number of packets to recover the source content. Then the source packets are re-encoded into a new set of LT packets that differ from the packets produced independently in the other nodes. This is made possible by the rate less property of LT codes, which allow for the generation of an infinite number of different packets. Decoding and encoding in the nodes however come at the price of increased complexity and delay. The network topology is however constructed such that minimal delays can be achieved. The streaming system is shown to be resilient to network dynamics with an increased throughput due to the rate less properties of the LT codes.

Decoding operations in the network nodes could be avoided and replaced by linear packet combinations. The work in [17] proposes to take benefit of the properties of Raptor codes that offer linear encoding and decoding complexity and rate less characteristics. Packets are encoded with Raptor codes at the servers. The network nodes then selectively combine packets when they have to compensate for packet losses and bandwidth variations.Such a system does not necessitate the use of large buffers in the nodes and the coding operations are kept very simple.

This solution is advantageous in terms of delay and complexity compared to methods that would implement Raptor decoding in the nodes, in a similar manner to LT decoding in. A rate allocation algorithm further determines the optimal source and channel rates so that the quality is maximized for the smallest capability client.

The resulting scheme is shown to be extremely robust to network variations.
Finally, network coding could also be used for recovering from errors in broadcast applications. When clients experience errors, packets retransmissions rapidly lead to bandwidth explosion as every client might request a different packet. Network coding is helpful in limiting the number of retransmissions since it replaces the retransmission of original packets by the transmission of packet combinations that can be decoded at the clients.

For example, the work in [18] studies the problem of broadcasting using network coding over one hop WiMAX networks. Network coding is applied whenever several packet losses are reported. Such a strategy is shown to be more efficient than state-of-the-art error resilient transmission schemes.

C. Prioritized network coding

Media streams are generally characterized by packets with different importance with respect to their contribution to the quality at the decoder. Network coding can adapt to this property by handling the packets according to their priority. Network coding based on Prioritized

Encoding Transmission (PET) principles has been initially proposed in [19], where data of high importance receives a high level of error protection by means of a proper arrangement of the data blocks in the encoding matrix. The PET algorithm is replaced by a MD-FEC scheme, which seeks for the distortion-minimal source and channel rate allocation for the given channel conditions.

Prioritized network coding is applied to scalable video streams. Data is segmented and interleaved in the coding matrix, in such a way that the base layer typically receives more redundancy bits than the enhancement layers. Classical network coding is then performed on packets of the same generation. The proposed scheme is shown to outperform other solutions based on either routing or routing with replication policies.

One could also achieve different levels of protection by changing the network coding scheme itself, where the coding operations are adapted to the importance of the packets. Improved data persistence is achieved due to the fact that the most important video data represents a combination of fewer source packets. The prioritized encoding problem is casted as inter-session network coding problem. In inter-session network coding, combinations of packets from different information sources are allowed when the clients simultaneously subscribe to all sources. The layered data can then be organized into multiple pipes that convey the network coding packets. The data from the most important layer typically flows into the first pipe. The second pipe transmits packet combinations from the first two layers. The other layers are arranged similarly, and network coding is applied on embedded sets of data. The packets from the most important layers are therefore used more frequently in the coding operations, which lead to a higher recovery probability in a progressive decoding scheme. Competition between the packets of the different layers is avoided in such a scheme, which however requires as many coding buffers as the number of quality levels. Unequal error protection can also be achieved by redefining the global encoding kernel (GEK) as proposed in [55]. This approach decomposes the network graph into connected line graphs with different coding operations. It optimizes the level of protection by solving an exhaustive search problem.

Prioritized coding can further be achieved by organizing the packets in multiple classes, depending on their importance. For example, the work in [20] addresses the problem of streaming wirelessly some H.264/AVC encoded video content. Packets are grouped in different classes, and frame dependencies are further taken into account for determining the optimal network coding operations for each class. The coding choices are determined locally in each node by estimating the number of innovative packets received by each client. However, the coding decisions are still complex to compute due to the high number of dependencies between packets. Media packets are grouped into classes of different importance and unequal protection is achieved at each node by varying the number of packets from each class that are used in the network coding algorithm. A low complexity greedy algorithm is used locally in each node in order to determine the best coding choice. The proposed scheme outperforms baseline network coding algorithms that do not take into account the importance of the packets for the delivery of layered media streams.

D. Adaptivity and opportunistic coding

Network coding can also be applied for media streaming on shared communication medium. When packet transmission can be overheard by multiple nodes, receivers could build up a buffer of packets that can be used to decode the successive packets. The senders can thus use some knowledge about the receiver status to optimize network coding operations and reduce the overall transmission costs. The COPE architecture has been presented in [21] for communication over wireless mesh networks. It introduces the concepts of listening and opportunistic coding.

The antennas listen to the broadcast channel and acquire packets that are stored temporarily in their buffers.

When the sender is informed about the receiver status, it could determine packets' combinations that maximize the probability of decoding for a maximum number of clients. For video transmission, the selection of network coding operations is based only on the maximization of the number of clients that can decode the packets. It is however suboptimal, since packets typically have different importance for the reconstructed video. The video quality is significantly improved by selecting the proper network codes that take into account the importance of the data as well as the timing deadlines.

A baseline scheme considers the importance of each packet and selects the most important packet for transmission from the top of the queue, ignoring packets that have been already been transmitted. A more advanced scheme uses the rate distortion optimization framework proposed in RaDiO [22] and incorporates it into the design of the network coding algorithm such that the expected distortion is minimized. Such a solution is shown to outperform simpler schemes that do not consider jointly packet deadlines and packet importance.

4.0. CONCLUSIONS

Network coding is an interesting paradigm that requests the network nodes to perform basic processing operations on packets in order to improve the throughput or the robustness of communication systems with network diversity. It has been shown in this paper that the theory of network coding finds application in media streaming. Multimedia communication could typically benefit from network coding in overlay networks or wireless mesh networks. Coding decisions could also be adapted to the packet importance or the state of the receivers in order to maximize the quality of service.Nevertheless, content-aware network coding and prioritization techniques have surely a strong potential and may improve the quality of the multimedia services and increase the robustness of the transmission. The choice of the right trade-off between delay, coding efficiency and complexity is still an open issue in network coding. The use of hybrid methods that exploit both the benefits of channel coding and network coding might provide interesting solutions to this compromise. Finally, It is expected that network coding will become a key technology for multimedia applications where the communication is performed in random networks with diversity.

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