Optimizing Network Coding Algorithms

for Multicast Applications

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Summary

Deviating from the archaic communication approach of treating information as a fluid moving through pipes, the concepts of Network Coding (NC) suggest that optimal throughput of a multicast network can be achieved by processing information at individual network nodes. However, existing challenges to harness the advantages of NC concepts for practical applications have prevented the development of NC into an effective solution to increase the performance of practical communication networks. In response, the research work presented in this thesis proposes cross-layer NC solutions to increase the network throughput of data multicast as well as video quality of video multicast applications.

First, three algorithms are presented to improve the throughput of NC enabled networks by minimizing the NC coefficient vector overhead, optimizing the NC redundancy allocation and improving the robustness of NC against bursty packet losses. Considering the fact that majority of network traffic occupies video, rest of the proposed NC algorithms are content-aware and are optimized for both data and video multicast applications. A set of content and network-aware optimization algorithms, which allocate redundancies for NC considering content properties as well as the network status, are proposed to efficiently multicast data and video across content delivery networks. Furthermore content and channel-aware joint channel and network coding algorithms are proposed to efficiently multicast data and video across wireless networks. Finally, the possibilities of performing joint source and network coding are explored to increase the robustness of high volume video multicast applications.

Extensive simulation studies indicate significant improvements with the proposed algorithms to increase the network throughput and video quality over related state-of-the-art solutions. Hence, it is envisaged that the proposed algorithms will contribute to the advancement of data and video multicast protocols in the future communication networks.

Key words: Network Coding, Practical Network Coding, Cross-Layer Optimization, Video Multicast

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List of Publications

Majority of the content in this thesis are included in peer reviewed publications. While some of the publications contain material generated from joint work, the contributions of this thesis are sole work of the author.

- C. de Alwis, H. Kodikara Arachchi, Anil Fernando, Varuna De Silva and Ahmet Kondoz, "On the Multicast of Delay Sensitive Data in Erasure Networks Using Practical Network Coding," in 2013 IEEE International Conference on Consumer Electronics (ICCE), 2013

- C. de Alwis, H. K. Arachchi, A. Fernando, and A. Kondoz, "Towards minimising the coefficient vector overhead in random linear Network Coding," in 2013 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), 2013, pp. 5127–5131.

- C. de Alwis, V. De Silva, H. K. Arachchi, A. Fernando, and A. Kondoz, "Network coding for error resilient transmission of stereoscopic 3D video," in 2013 IEEE International Conference on Communications Workshops (ICC), 2013, pp. 607–611.

- C. de Alwis, H. Kodikara Arachchi, V. De Silva, A. Fernando, and A. Kondoz, "Robust video communication using Random Linear Network Coding with Pre-Coding and Interleaving," in 2012 19th IEEE International Conference on Image Processing (ICIP), 2012, pp. 2269–2272.

- Politis, V. De Silva, S. Dogan, C. de Alwis, T. Dagiuklas, and A. M. Kondoz, "Objective measurement of transmission losses and compression artefacts in stereoscopic 3D video," in 2013 IEEE International Conference on Communications Workshops (ICC), 2013, pp. 667–671.

- E. Danish, D.V.S.X. De Silva, W.A.C. Fernando, C. De Alwis, and A.M. Kondoz, "Content Aware Resource Allocation in OFDM Systems for Energy Efficient Video Transmission," in 2013 IEEE International Conference on Consumer Electronics (ICCE), 2013

Glossary of Terms

3D	Three-Dimensional
ARQ	Automatic Repeat reQuest
AVC	Advanced Video Coding
DROVMM	Distributed Robust Optimization for Scalable Video Multirate Multicast
ETX	Expected transmission count
FEC	Forward Error Correction
FF	Freedom Feedback
GOP	Group of Pictures
HD	High Definition
JSVM	Joint Scalable Video Model
LDPC	Low Density Parity Check
NC	Network Coding
NC-MGM	Network Coding with Multi-Generation Mixing
OSI	Open Systems Interconnection model
PNC	Practical Network Coding
PNCPCI	Practical Network Coding with Pre-Coding and Interleaving
PSNR	Peak Signal-to-Noise Ratio
QP	Quantization Parameter
RLNC	Random Linear Network Coding
SCM	Standard Coefficient Matrix
SDN	Software Defined Networking
SVC	Scalable Video coding

TF	Traditional Feedback
VCU	Video Coding Units
VoD	Video on Demand
WMN	Wireless Mesh Networks
WSN	Wireless Sensor Networks

Chapter 1

1 Introduction

1.1 Preamble

The annual global network traffic volume has been increasing exponentially with the popularity of the Internet and the vast number of emerging online services. It is envisaged that the annual global network traffic will reach the zettabyte threshold by the end of 2015 and continue to grow up to 1.4 zettabytes by 2017 [1]. The majority (80% - 90%) of network traffic is expected to contain some form of video, such as Internet television, Video on Demand (VoD), Internet video, peer-to-peer video, etc. considering the increasing demand for high volume video services. However, from a network perspective, the scarcity of network resources poses a significant challenge to the efficient delivery of high volume data and video. This induces a plethora of research opportunities to explore efficient means of delivering high volume data and video across networks by overcoming practical concerns such as bandwidth limitations, packet losses, packet delays, etc.

In response, numerous research contributions have demonstrated the provision for multi-fold improvements in network throughput. Network Coding (NC) is one of the promising technologies that has proven its ability to enhance not only the throughput but also robustness and reliability of networks [2]–[7]. In spite of the archaic approach of treating information as a fluid moving through pipes, the concepts of NC reveal that optimal throughput of a multicast network can be achieved by processing information at individual network nodes [2]. In doing so, NC is capable of cleverly utilizing the multiple paths available between the source and destination nodes. However, a number of under-explored technological challenges remain to be addressed in order to effectively utilize NC technologies for data and video multicast applications.

This thesis proposes intelligent channel, network and content-aware NC algorithms to facilitate high volume data and video multicast applications, which will increase the performance of scarce network resources to pave the path for a better connected world.

1.2 Problem Statement and Research Objectives

Owing to the fact that NC is a viable solution to efficiently utilize network resources, developing NC protocols have already become a prominent theme in the state-of-the-art communication research. Despite the numerous approaches followed to develop NC algorithms, the existing limitations have prevented the development of NC into an effective solution that caters to the future demands of communication networks.

For instance, the performance of existing NC algorithms is significantly reduced by the NC coefficient overhead [8]. Minimizing this overhead will in return increases the throughput of NC enabled networks. Furthermore, improving the robustness of existing NC algorithms by considering various issues encountered under practical networks can improve the feasibility of deploying NC techniques in practical communication networks. This includes improving the Forward Error Correction (FEC) capabilities to optimize NC algorithms for delay sensitive multicast applications and enhancing the robustness of NC against bursty packet losses, which are commonly occurred in practical networks. In addition, adopting NC algorithms to be used under extremely high packet loss rates observed in wireless networks can increase the throughput and robustness of state-of-the-art wireless network infrastructure.

Considering the fact that the majority of network traffic contains some form of video, as elaborated in Section 1.1, improving the performance of video multicast applications will in return improve the overall network throughput. This can be performed by exploiting the unequal importance of video packets in allocating network resources to design content and network-aware NC algorithms. What is more, by means of harmoniously allocating network resources for channel and network coding, joint source-channel and network coding algorithms can be designed to improve the performance of high volume data and video multicast applications.

In response, the research work presented in this thesis proposes cross-layer NC algorithms, which efficiently utilize network resources to increase the performance of data and video multicast applications.

The main objectives of this thesis are outlined below:

- Objective 1 : Design NC algorithms to increase the network throughput by minimizing the NC coefficient overhead and improving the robustness of a transmission against packet losses.

- Objective 2 : Propose content and network aware NC algorithms to increase the performance of data and video multicast applications.

- Objective 3 : Jointly optimize channel coding and NC by considering content properties and channel status to deliver data and video multicast services across wireless networks.

1.3 Thesis Contributions

The work presented in this thesis is performed to meet the objectives outlined in Section 1.2. Accordingly, six distinct contributions are made in this thesis as listed below:

- **Contribution 1 : Optimized Practical Network Coding (PNC) algorithms for efficient multicast** (Chapter 3)

  Existing PNC algorithms incur a significant overhead in conveying coding information to sink nodes in order to assist the decoding of NC packets [8], [9]. This reduces the amount of payload per NC packet. Addressing this issue, a novel algorithm is proposed to minimize the coefficient vector overhead of PNC, which in return increases the network throughput.

  Furthermore, it should be noted that the end-to-end probability of an NC packets being delivered to a sink node in a PNC enabled network cannot be defined since each node transmits newly generated packet combinations [10]. Hence, existing PNC solutions are not capable of sensing the end-to-end network status to allocate redundancies for PNC such that the robustness of a multicast is enhanced in a bandwidth efficient manner. In response, a novel method is presented to estimate the end-to-end probability of an NC packet reaching a sink node in a multicast network and apply redundancies in a network-aware fashion for delay sensitive

data multicast applications such that either the required packet delivery probability or the optimal throughput is achieved.

In addition, it is observed that bursty packet losses increase the possibility of sink nodes not receiving sufficient packet combinations to decode groups of NC packets. As a solution, Practical Network Coding with Pre-Coding and Interleaving (PNCPCI) is proposed to incorporate the concepts of interleaving and pre-coding to PNC such that the robustness against bursty packet losses is increased. Random packet losses are recovered during the process of decoding NC packet combinations. Bursty packet losses are spread after deinterleaving and recovered while decoding pre-coded packet combinations.

- **Contribution 4 : Content and network-aware NC for delay sensitive data and video multicast applications** (Chapter 4)

    Existing algorithms are not optimal in allocating redundancies for NC to efficiently deliver delay sensitive data and video multicast services across networks. In response, a novel NC algorithm is proposed, which efficiently utilizes cross-layer information, to improve the packet delivery probability of a delay sensitive data multicast. Furthermore, the proposed algorithm is optimized to increase the received video quality of video multicast by means of optimizing the redundancy allocation for PNC by considering both the network status and content properties. In addition, the probability of continuous video playback is increased at all users in the network.

- **Contribution 5 : Content and channel-aware channel coding and NC for the efficient multicast of data and video over wireless networks** (Chapter 5)

    Wireless networks exhibit extremely high packet loss rates due to significant fading present in wireless channels. Hence, a novel algorithm is presented to increase the robustness of bandwidth sensitive data and delay sensitive video multicast applications across wireless networks. Each network node is capable of independently allocating redundancies for channel coding and network coding taking in to account the content properties, channel bandwidth and channel status to improve the end-to-end performance. Both bandwidth data and video multicast applications have been considered in optimizing the proposed algorithm. For data

multicast applications, redundancies are allocated at each network node in such a way that the total amount of redundant bits transmitted is minimized. As for video multicast applications, redundancies are allocated considering the priority of video packets such that the probability of delivering high priority video packets is increased. This not only increases the probability of continuous video playback but also improves the received video quality.

- **Contribution 6 : Redundancy allocation between source coding and network coding for high volume video multicast applications** (Chapter 6)

  The opportunities of harmoniously allocating video redundancies and PNC redundancies in performing joint source and network coding to increase the error resilience of high volume video multicast are not investigated in state-of-the-art NC research. Addressing this, an extensive simulation study is performed to identify efficient means of allocating redundancies between video coding and PNC to increase the robustness of a high volume video multicast across lossy networks. Furthermore, the requirement of optimizing existing video coding algorithms for PNC is pointed out to design joint source-channel and network coding algorithms to increase the network resource utilization in high volume video multicast applications.

1.4 Thesis Outline

This thesis is organized in to seven chapters. The first chapter, which includes this section, provides an introduction to the research problem and summarizes research objectives and contributions. Rest of the chapters is organized as follows. A brief review of existing literature related to the work presented in this thesis is presented in the second chapter. Chapters 3, 4, 5 and 6 correspond to the technical contributions presented in this thesis. These chapters introduce the related research problem and present the proposed solutions along with simulation results, compared with relevant state-of-the-art algorithms. Finally, the conclusions and future work relevant to the proposed work is described in chapter 7.

Chapter 2

2 Network Coding for Multicasting

2.1 Introduction

This chapter overviews the theoretical background of NC techniques and elaborates state-of-the-art research work related to the technical contributions presented in this thesis. The concept of network coding or employing coding at network nodes is introduced by Ahlswede *et al.* to transmit information reaching the network capacity [2], as defined by the max-flow min-cut theorem. It should be noted that the max-flow min-cut theorem states that the maximum amount of flow that can pass from the source node to a sink node in a network is equal to the minimum capacity between the two nodes.

The source node of an NC enabled network transmits combinations of information packets across a network. Intermediate nodes that receive multiple packet combinations, recombine the received packet combinations and transmit newly generated packet combinations. Sink nodes recover the original packets upon receiving sufficient packet combinations to decode network coded packets. Consider a butterfly network as illustrated in Figure 2-1. Node 0 is the source node, which is required to multicast packets *a* and *b* to the sink nodes, node 5 and node 6. The remaining nodes are intermediate network nodes. All the links are unity capacity links, i.e., each link is only capable of transmitting a single packet during a considered time interval. Due to the bandwidth constraint in the link between node 3 and node 4, the maximum multicast rate becomes 1.5 packets per time interval. However, when network coding is applied by performing the *xor* operation on the packets received by node 3,as illustrated in Figure 2-2, the maximum multicast rate increases to 2 packets per time interval.

6
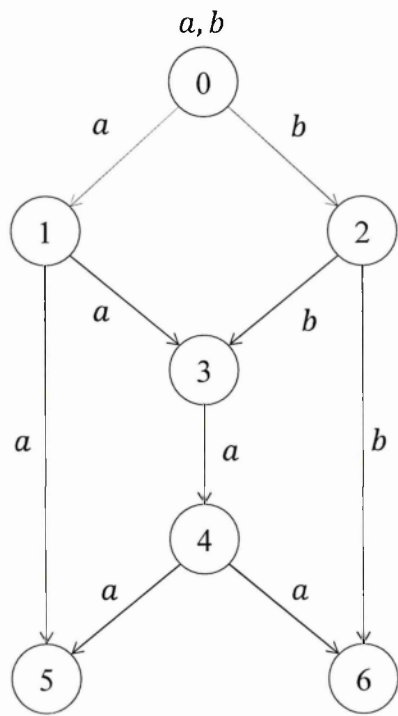
Figure 2-1 Traditional routing in a butterfly network topology
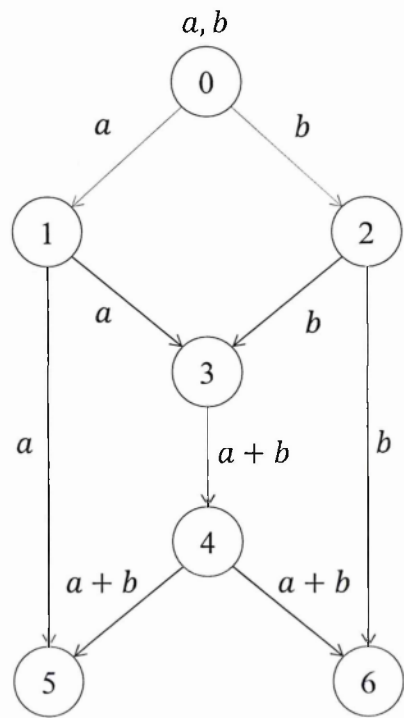
Figure 2-2 Network coding implemented in a butterfly network topology

Li et al. proved that performing linear coding, which is one of the simplest coding schemes, to generate NC packet combinations is sufficient to achieve the network capacity as defined by the max-flow min-cut theorem [4]. Furthermore, Koetter and Medard presented an algebraic framework for NC, connecting sets of polynomial equations and their solutions to networks [11]. This work extended the NC framework to be suitable for arbitrary network structures, where each node uses random coefficients selected from a finite field to generate linear combinations of received packets. A Random Linear Network Coding (RLNC) approach to multicast is presented in [12] to enable network nodes to independently select random coefficients from a finite field to perform linear NC, thereby facilitating distributed networking operations and networks with dynamically varying connections.

2.2 Random Linear Network Coding

Random linear network coding uses a randomized strategy of performing network coding to achieve network capacity in an arbitrary network structure [11]–[22]. All nodes other than the sink nodes, which receive multiple packet combinations perform random linear mappings from input packets to output packets over a finite field. Sink nodes can decode the original packets by knowing the overall linear combination of incoming transmissions. This information can be sent with each transmission block or packet as a vector of coefficients. These coefficients are updated at each network node by applying the same linear mappings as to the packet combinations. An example of RLNC implemented in a butterfly network, as described in Section 2.1, is illustrated in Figure 2-3. Accordingly, the source node, node 0, needs to multicast packets a, b to the sink nodes, node 5 and node 6. The source node generates two linearly independent packet combinations using randomly generated coefficients c_1, c_2, c_3 and c_4 selected from a certain finite field. These packet combinations are transmitted along with the respective coefficients used to generate the particular packet combination. Node 3, which receives two packet combinations, generates new packet combinations using randomly generated coefficients c_5 and c_6 selected from the same finite field. Furthermore, the coefficients of received packet combinations are updated using the same randomly generated coefficients c_5 and c_6. Then, node 3 transmits the newly generated packet combinations with the updated set of coefficients. Since node 4 only receives a single packet combination, it forwards the received packet combination along with the NC coefficients to Nodes 5 and

6. Using the received packet combinations and coefficient vectors, the sink nodes, node 5 and node 6, can recover the original packet combinations a and b.

2.3 Practical Network Coding

Practical Network Coding [23] develops random linear network coding algorithms to increase the robustness against packet losses, packet delays and topology changes by,

a, b

$c_1.a + c_2.b$

$c_3.a + c_4.b$

$c_5.(c_1.a + c_2.b) + c_6.(c_3.a + c_4.b)$
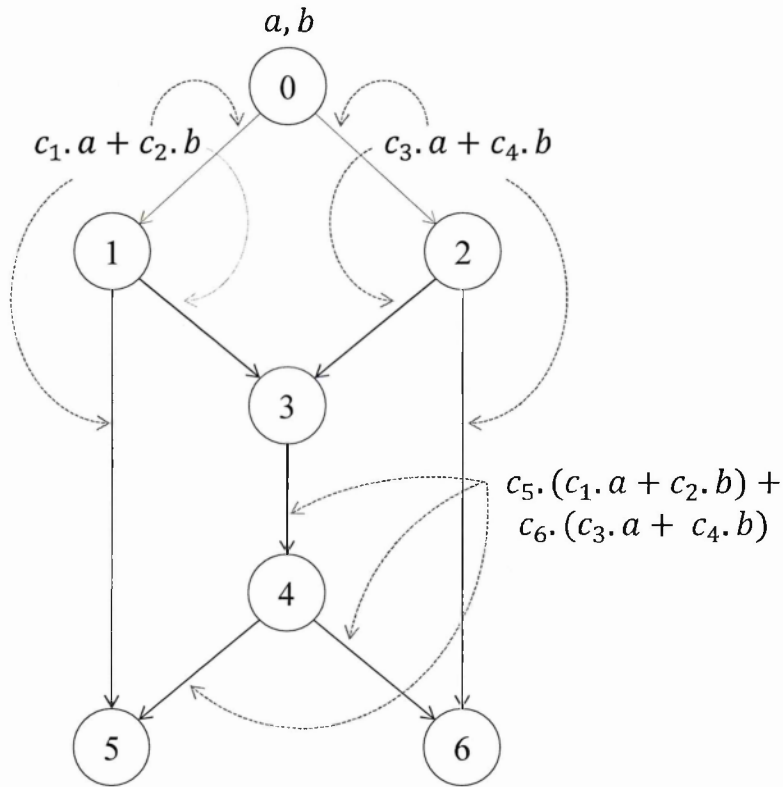
Figure 2-3 Random linear network coding implemented in a butterfly network topology

1. Buffering packet combinations at each network node to synchronize arbitrary packet arrivals and departures at each node.

2. Performing random encoding to deal with varying numbers of packets in the buffer.

3. Introducing a packet format that includes global encoding vectors to provide the sink nodes with sufficient information to decode network coded packets.

In doing so, PNC achieves throughput approaching the network capacity defined by the max-flow min-cut theorem.

Consider a PNC enabled multicast network. The source node linearly combines n source packets to generate $m(m \geq n)$ linear combinations. These m packet combinations belong to the same group of packets identified as a generation. The $m - n$ number of additional packet combinations are transmitted in order to ensure the successful reception of packet combinations by the sink nodes in case of packet losses. However, it should be noted that the additional packets introduces $\frac{(m-n)}{m}$ % of redundancies to the network. Let $[X]_{n \times s}$ be n source packets, where, each source packet is a $1 \times s$ matrix of symbols from a Galois field of size $2^F(F$ is the order of the Galois field). A $[C]_{m \times n}$ matrix with rank n is generated at the source using coefficients from the same Galois field of 2^F. m packet output packet combinations, $[Y]_{m \times s}$ are generated using,

$$[Y]_{m \times s} = [C]_{m \times n} \cdot [X]_{n \times s}$$

(2-1)

Subsequently, each packet combination is transmitted from the source node together with the respective set of coefficients used to generate the particular packet combination. The rows of the composite matrix $[[C]_{m \times n} \ [Y]_{m \times s}]$ represents the m output packets.

Intermediate nodes, which receive multiple packet combinations from the same generation linearly recombine the received packet combinations to generate k packet combinations, where, $k(\geq 1)$ is the number of output links from the particular node. Let a particular node receive $l(\geq 1)$ combinations from the same generation. The received packets can be represented as $[[C']_{l \times n} \ [Y']_{l \times s}]$, where $[C']_{l \times n}$ are the coefficients used to generate the received $[Y']_{l \times s}$ packet combinations. The node will generate a coding matrix $[\hat{C}]_{k \times l}$ where each element is a randomly selected coefficient from the same Galois field of 2^F. The k output combinations are generated as,

$$[Y'']_{k \times s} = [\hat{C}]_{k \times l} \cdot [Y']_{l \times s}$$

(2-2)

The coefficient matrix is updated accordingly by,

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$$[C'']_{k \times n} = [\hat{C}]_{k \times l} \cdot [C']_{l \times n} \tag{2-3}$$

Each row of $[[C'']_{k \times n} \ [Y'']_{k \times s}]$ represents the k output packets transmitted across the k output links of the node.

Sink nodes collect packet combinations, which increase the rank of the received coefficient matrix (innovative packets) of a particular generation, and discard any linearly dependent packet combinations (non-innovative packets). Once the rank of a received coefficient matrix becomes n, sink nodes can decode the network coded packets belonging to the particular generation. If the n innovative packets are rows of $[[\hat{C}]_{n \times n} \ [\hat{Y}]_{n \times s}]$, where $[\hat{C}]_{n \times n}$ are the coefficients relevant to generate the received $[\hat{Y}]_{n \times s}$ packet combinations, the network coded packets are decoded by,

$$[\hat{X}]_{n \times s} = [\hat{C}]_{n \times n}^{-1} \cdot [\hat{Y}]_{n \times s} \tag{2-4}$$

2.3.1 Compression of Coding Vectors in Practical Network Coding

PNC appends the coefficient vectors in front of the network coded packets to provide sink nodes with the information on how packets are encoded. These coefficients are updated at each node where network coding is performed. Practical network coding splits the number of packets to be coded using random linear network coding in to fixed sized groups called generations. The number of packets belonging to a generation is called the generation size. However, the transmission of coefficient vectors comes at the cost of additional bandwidth. Consider a network coding scheme which uses a generation size of 100 and a Galois field. This scheme requires coding vectors to accommodate 200 bytes per packet. Even considering a larger packet size of 1400 bytes, over 14% of payload data is required to transmit coefficient vectors. Furthermore, increasing the generation size or using a larger finite field of Galois field results in increasing the coefficient vector size.

As a solution, a method named Subspace coding is introduced in [24], which, does not require transmitting coefficient vectors. Subspace coding uses a particular subspace

selected by the information source to convey information. Sinks can decode the coded information by deciding the particular subspace. However, for larger packet lengths, the information rate achieved by subspace coding becomes the same as sending coefficient vectors [9]. Furthermore, designing subspace codes also becomes challenging for a multi-source networks [8]. An alternative approach to compress network coding vectors is proposed in [8]. The authors argue that the packets received at information sinks may only contain linear combinations of several source packets, which represents a fraction of the total number of source packets. Therefore, the number of coefficient vectors transmitted is limited by limiting the number of packets that are allowed to be combined in a network. However, depending on the network structure and the generation size, the difference between the number of source packets included in a linear combination received by a sink and the number of source packets, maybe negligible. This will significantly reduce the gain achieved by the compressed coding vectors algorithm proposed in [8]. In addition, limiting the number of packets being combined may decrease the probability of a redundant combination being useful for multiple sinks in a network with packet losses.

2.3.2 Feedback Schemes for Practical Network Coding

PNC linearly combines a group of packets belonging to a generation and transmits the linear combinations across a network. Intermediate network nodes recombine the packet combinations belonging to the particular generation and transmits newly generated packet combinations. After receiving sufficient number of linear packet combinations, sink nodes are able to decode the original packets belonging to the particular generation. However, since multiple source packets are encoded and decoded as a group and intermediate nodes independently re-encode network coded packets, it is not possible for sink nodes to acknowledge individual source packets before they are decoded. Qu et al. presents a comprehensive comparison of feedback schemes for network coding in [25]. Accordingly, the feedback mechanisms for NC are divided in to two categories, namely Traditional Feedback (TF) and Freedom Feedback (FF).

In TF for NC, a source node keeps transmitting packets combinations from a generation until it is acknowledged by a sink node stating that it has received sufficient linearly independent NC packet combinations to decode the particular NC generation [26], [27]. However, TF may cause excess transmissions, which degrades the network performance.

In response, FF for NC is proposed to communicate with the source indicating how many more linearly independent NC packets are required to decode a particular generation [28]–[30]. In doing so, FF reduces both the queue size and the decoding delay. However, FF requires the source and the sink nodes to continuously exchange feedback information. Any delayed feedback information will cause prolonged waiting times at the source node. According to [25], FF achieves better delay performance and requires fewer transmissions under small transmission intervals compared to TF. In contrast, at larger transmission intervals, the performance of TF becomes better compared to FF in terms of decoding delay while FF requires almost the same total number of transmissions.

However, both the TF and FF schemes exhibit increased decoding delays and need more transmissions as the packet loss rate in links increase. In addition, both TF and FF schemes are less suitable for delay sensitive data multicast in large scale multi-hop networks due to the delays incorporated being significant.

2.4 Network Coding with Multi-Generation Mixing

PNC encodes packets belonging a generation and transmits the linear combinations across the network. In contrast, Network Coding with Multi-Generation Mixing (NC-MGM) groups generations in to mixing sets and mixes packets belonging to a particular set in order to increase the robustness against packet losses [31].

Consider an NC-MGM enabled multicast network. Let the size of a mixing set be z generations. Each generation in the mixing set has a position index. The position index of a generation indicates its relative position within the mixing set. Let the position index, d $(1 \leq d \leq z)$ of the first generation in the mixing be one and the last generation in the mixing set be z. The source node linearly combines the first $d.n$ number of source packets of a mixing set to generate $d.m$ number of linear combinations, where $m \geq n$, as illustrated in Figure 2-4. The $z.(m-n)$ additional packet combinations from a mixing set are transmitted in order to ensure the successful reception of packet combinations by the sink nodes in case of packet losses. Let $[X_d]_{n \times s}$ be n source packets belonging to the dth generation in the mixing set. Each source packet is a $1 \times s$ matrix of symbols from a Galois field of size 2^F (F is the order of the Galois field). A $[C]_{m \times (d.n)}$ matrix with rank $d.n$ is generated at the source using coefficients from the same Galois field of 2^F. The m output packet combinations from the dth mixing set are generated such that,
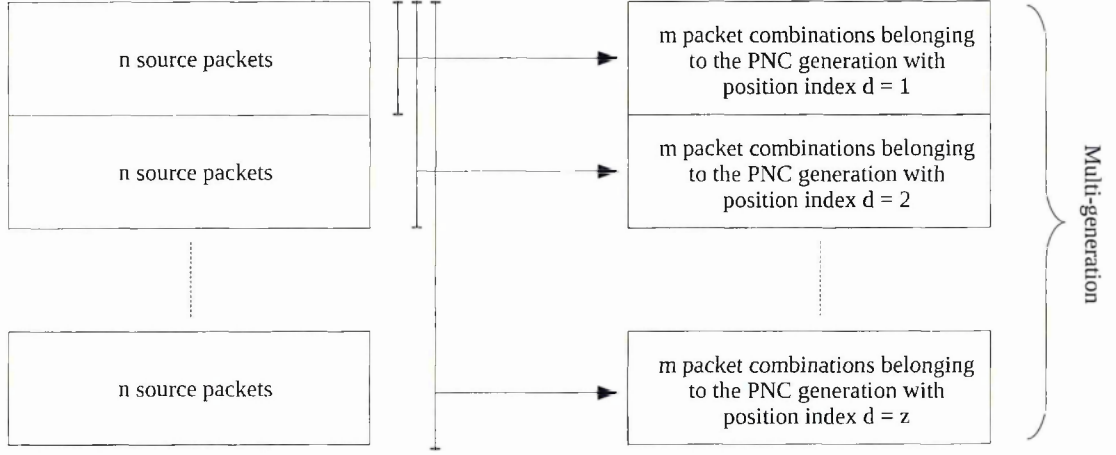
Figure 2-4 Practical network coding with multi-generation mixing

$$[Y]_{m\times s} = [C]_{m\times(d.n)} \cdot [X]_{(d.n)\times s}$$

(2-5)

Similar to the case of PNC, these m packet combinations belonging to the $d.z^{th}$ NC generation are transmitted from the source node together with the respective set of coefficients.

The operation of intermediate nodes is similar to PNC, as they encode packet combinations belonging to the same NC generation. Sink nodes collect innovative packet combinations from a particular mixing set. When the rank of a received coefficient matrix from the first d generations of a mixing set becomes $d.n$, sink nodes decode the $d.n$ NC-MGM packets. Let the $d.n$ innovative packets be $\left[[\hat{C}]_{(d.n)\times(d.n)} [\hat{Y}]_{(d.n)\times s}\right]$, where $[\hat{C}]_{(d.n)\times(d.n)}$ are the coefficients relevant to generate the received $[\hat{Y}]_{(d.n)\times s}$ packet combinations, the network coded packets are decoded by,

$$[\hat{X}]_{d.n\times s} = [\hat{C}]^{-1}_{(d.n)\times(d.n)} \cdot [\hat{Y}]_{d.n\times s}$$

(2-6)

It should be noted that in case of a packet loss, the $(m - n)$ number of redundant packets generated from generation d in a mixing set are useful in decoding any of the $1 \dots d$ generations in the particular mixing set. This increases the versatility of redundant packets

as they can be used to recover more than one generation in case of a packet loss, which in return increases the robustness against packet losses.

2.5 Network Coding for Wireless Mesh Networks

Wireless Mesh Networks (WMNs) are increasingly being deployed to provide cost efficient and low maintenance internet access [26], [32]–[34]. However, due to fading, wireless links in WMNs exhibit extremely high packet loss rates. For instance, half of the links in Roofnet [32] exhibit over 30% packet loss rates. Techniques of using NC based multicast protocols to increase the throughput and robustness of a data multicast across WMNs is proposed in [26], [34]–[64]. These technologies allow network nodes to sense the forward links to estimate the number of coded packets they should transmit in order to increase the throughput and reliability of a data multicast.

PNC [65] introduces the concept of performing RLNC for groups of packets called generations to extend the usability of network coding in realistic networks with packet erasures, delays and topology changes. In addition, practical network coding allows transmitting redundant network coded packets as a FEC mechanism to increase the probability of decoding network coded packet generations amidst packet erasures. These redundant network coded packets perform FEC more efficiently than traditional Automatic Repeat reQuest (ARQ) based schemes to combat packet erasures [66]. Combining RLNC together with opportunistic routing, MORE [26] exploits the opportunism inherent in the wireless medium. In doing so, the throughput is significantly increased compared to traditional routing and prior work on opportunistic routing. Pacifier [26] further improves the throughput of a data multicast across a WMN by integrating tree-based opportunistic routing, network coding, source-rate limiting and round-robin batching. However, because of the excessive redundancy overhead of existing algorithms to achieve 100% packet delivery, they are not optimal for bandwidth sensitive applications. Furthermore, these algorithms are content agnostic. Nevertheless, in a video multicast, it is essential to deliver at least the most important packets in a timely manner to provide an uninterrupted video service at an acceptable quality.

2.6 Network Coding for Video Communication

The increased throughput and robustness against packet losses qualify NC to efficiently transmit high volume video content across practical networks [21], [67]–[93]. However, it should be noted that the direct application of NC algorithms is not optimal for efficient video communication.

It is common knowledge that different video packets within the same video stream have an unequal importance in contributing to distortion depending on the video content, video encoding, or play out deadlines. This unequal importance of packets should be considered to facilitate prioritized transmission of video over an NC enabled network. One such solution is to provide an unequal error protection with NC using priority encoding transmission (PET) [21], [94]–[97]. However, these methods are not capable of efficiently utilizing bandwidth resources to multicast video across packet lossy networks. Two of the state-of-the-art solutions, which efficiently utilize network resources to deliver video packets having a higher priority whith a higher probability are presented in [71] and [98]. The algorithm proposed in [71] groups video packets in to generations depending on their priority and adds more redundancies for generations consisting of high priority video packets. Alternatively, [98] proposes using NC-MGM for video transmission by adding high priority video packets to generations with a higher probability of decoding. These two algorithms are elaborated in Sections 2.6.1 and 2.6.2.

2.6.1 Robust Video Multicast with Joint Network Coding and Video Interleaving

An NC based cross-layer approach to perform robust video multicast in erasure networks is proposed in [71]. An overview of this method is presented in Figure 2-5. In this method, Video Coding Units (VCU) are partitioned into several priority levels considering scalability properties of H.264/SVC video. Next, the VCUs are interleaved with each other such that video packets belonging to the same priority level of multiple VCUs are grouped together. Subsequently, PNC is performed for each of these groups generating redundant packet combinations for each of the NC generations. These redundant packet combinations are generated providing an unequal error protection for

each generation such that higher redundancies are applied to protect high priority video data.
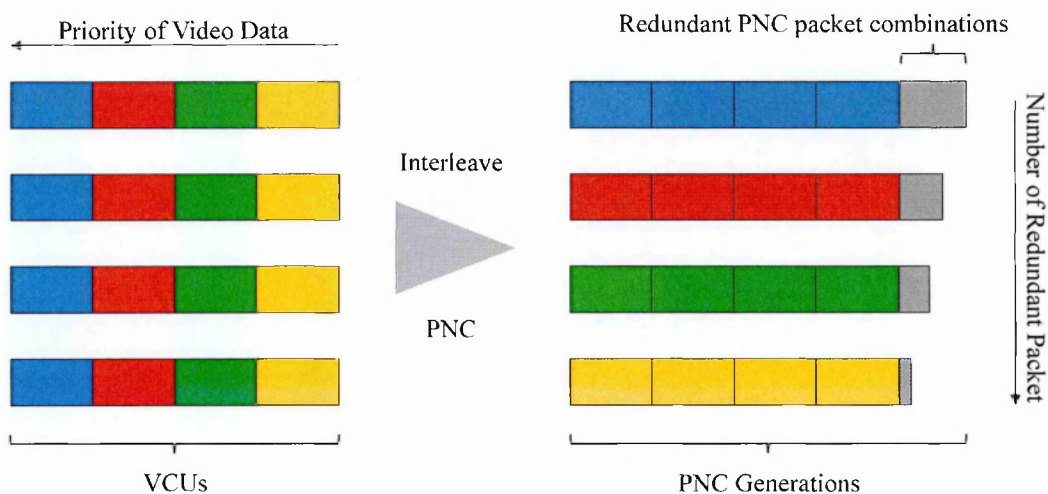
Figure 2-5 Abilene network topology used to simulate the proposed algorithm

2.6.2 Scalable Video Multicast using Network Coding with Multi-Generation Mixing

NC-MGM mixes packets from multiple generations to improve the robustness against packet losses. Furthermore, as explained in Section 2.4, the redundant packets generated from a generation in a mixing set are useful in decoding the considered generation and any of the preceding generations in the mixing set in case of packet losses. Therefore, generations that have a lower position index in a mixing set in a multi-generation exhibit a higher decoding probability when transmitted across a network exhibiting packet losses. Exploiting this characteristic, Halloush *et al.* proposed to use NC-MGM for scalable video multicast by allocating video packets to generations in a mixing set, considering the priority of the scalability layers. Consider a scalable video with z scalability layers. Video packets belonging to z scalability layers are allocated in to a multi-generation with z generations. This is performed such that the video packets belonging to the hth scalability layer is allocated to the generation having the position index of $z - h + 1$ in the multi-generation, as illustrated in Figure 2-6. Accordingly, the first scalability layer, which has the highest priority in decoding the video sequence at the sink node, is allocated to the generation having the position index 1 in the mixing set.
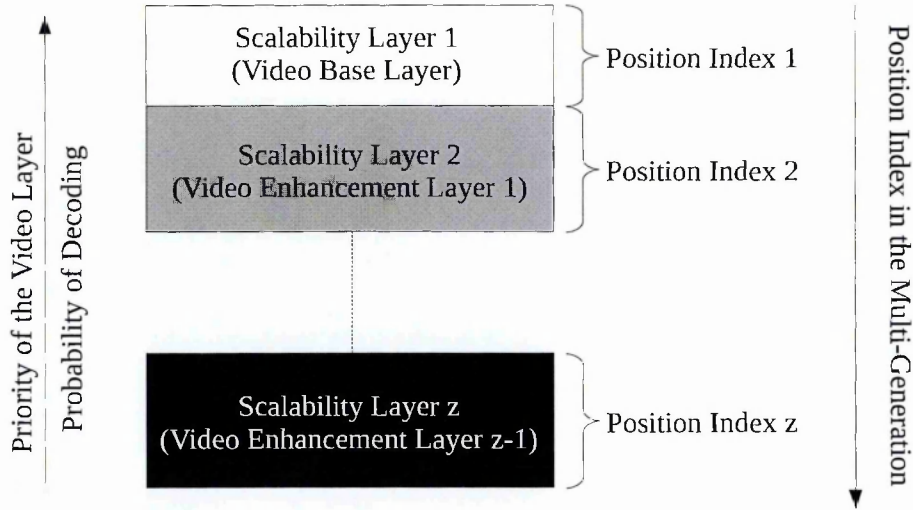
Figure 2-6 Abilene network topology used to simulate the proposed algorithm

By means of using NC-MGM for scalable video multicast, high priority scalability layers can be delivered at a higher probability. As a result, the received video quality of a scalable video multicast across a network exhibiting packet losses can be increased.

2.6.3   Network Coding for Video Multicast across Wireless Mesh Networks

Bandwidth sensitive content multicast applications, such as software updates required by many nodes across a network, require a 100% packet delivery ratio. For these kinds of applications, transmitting coded packets until receivers acknowledge the receiving of sufficient coded packets to decode a network coded packet generation, such as in [26], [34] is acceptable. Nevertheless, in a video multicast, it is essential to deliver at least the most important packets in a timely manner to provide an uninterrupted video service at an acceptable quality.

A delay sensitive video multicast such as internet television requires the timely delivery of video packets to ensure continuous video playback. Hence, it is not possible for a source node to delay the transmission of coded packets until all receivers acknowledge the reception of sufficient coded packets to decode the preceding network coded packet generation. Therefore, video packets are transmitted with redundancies added at the source node as a measure of FEC. In addition, a video multicast does not essentially require a 100% packet delivery rate as video decoders are capable of concealing missing

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video packets. Moreover, depending on the content, different video packets have a different impact in recovering the original video sequence. A distributed scheme to sense network conditions and adaptively transmit scalable video layers to optimize the received video quality is proposed in [99]. This algorithm uses the information from all network nodes to jointly optimize the overall video quality for a scalable video multi-rate multicast using network coding. In doing so, each scalable layer is tailored in an incremental order to find jointly optimal multicast paths and associated rates. The usage of network coding in the above algorithm increases the network throughput and enhances the robustness of a video transmission against packet losses. Nevertheless, the above algorithm allocates redundancies at the source node and the redundancy rate is kept constant throughout the multicast. Furthermore, the redundancy allocation in this algorithm is not optimal because it doesn't consider either the priority levels of scalability layers of a video or the status of individual links.

2.7 Summary

This chapter overviews the theoretical background of NC and state-of-the-art NC algorithms related to the presented work. The concept of NC is proposed to multicast data at the network capacity, as defined by the max-flow-min-cut theorem. RLNC, PNC and NC-MGM develop NC to be used in practical networks that exhibit packet losses, packet delays, random topology changes, etc. However, the concepts are far from practical, leaving numerous opportunities to be investigated to deploy NC technologies in practical networks. This includes minimizing the NC overhead and increasing the robustness of NC against packet losses. Furthermore, owing to the fact that the majority of network traffic contains video, NC concepts can be optimized to efficiently utilize network resources in video multicast applications. However, existing solutions are not optimal in efficiently utilizing network resources for high volume data and video multicast applications.

Chapter 3

3 Optimized Practical Network Coding Algorithms for Efficient Multicast

3.1 Introduction

Chapter 2 in this thesis highlights the existing PNC algorithms and points out the shortcomings that remain to be addressed. In response, Chapter 3 proposes three novels algorithms to improve the throughput of PNC in a multicast network. The rest of this chapter is organized as follows. Section 3.2 proposes an approach to minimize the coefficient vector overhead in PNC, which in return improves the network code rate by increasing the network throughput. Subsequently, Section 3.3 proposes a method of estimating the end to end probability of an NC packet being delivered to a sink node and thereby optimizes the allocation of redundancies to enhance the FEC capability of PNC. In addition, an algorithm to improve the robustness of PNC against packet loss bursts is presented in Section 3.4. Finally, Section 3.5 summarizes the contributions of the chapter.

3.2 Minimizing the Coefficient Vector Overhead in Practical Network Coding

Existing algorithms to minimize the coefficient vector overhead in PNC are not optimal as explained in Section 2.3.1. In response, this section proposes a novel approach to minimize the coefficient vector overhead of PNC. The proposed algorithm is overviewed in Figure 3-1. Accordingly, the proposed approach linearly combines source packets using a set of coefficient vectors, which are also known to sink nodes. The matrix containing coefficient vectors is identified by the term coefficient matrix. Instead of sending the coefficient vectors themselves, coefficient information or information

indicating which row/rows of the coefficient matrix are used to generate the particular linear combination, is sent along with the coded packet. Coefficient information is further compressed using a lossless compression scheme before transmitting. Intermediate nodes decompress the compressed coefficient information, combine received packet combinations, update and compress coefficient information and send the new packet combination along with updated coefficient information to the next hops. Sink nodes decompress the coefficient information of received packets, replaces the coefficient information with coefficient vectors appropriately and decodes the NC generations.
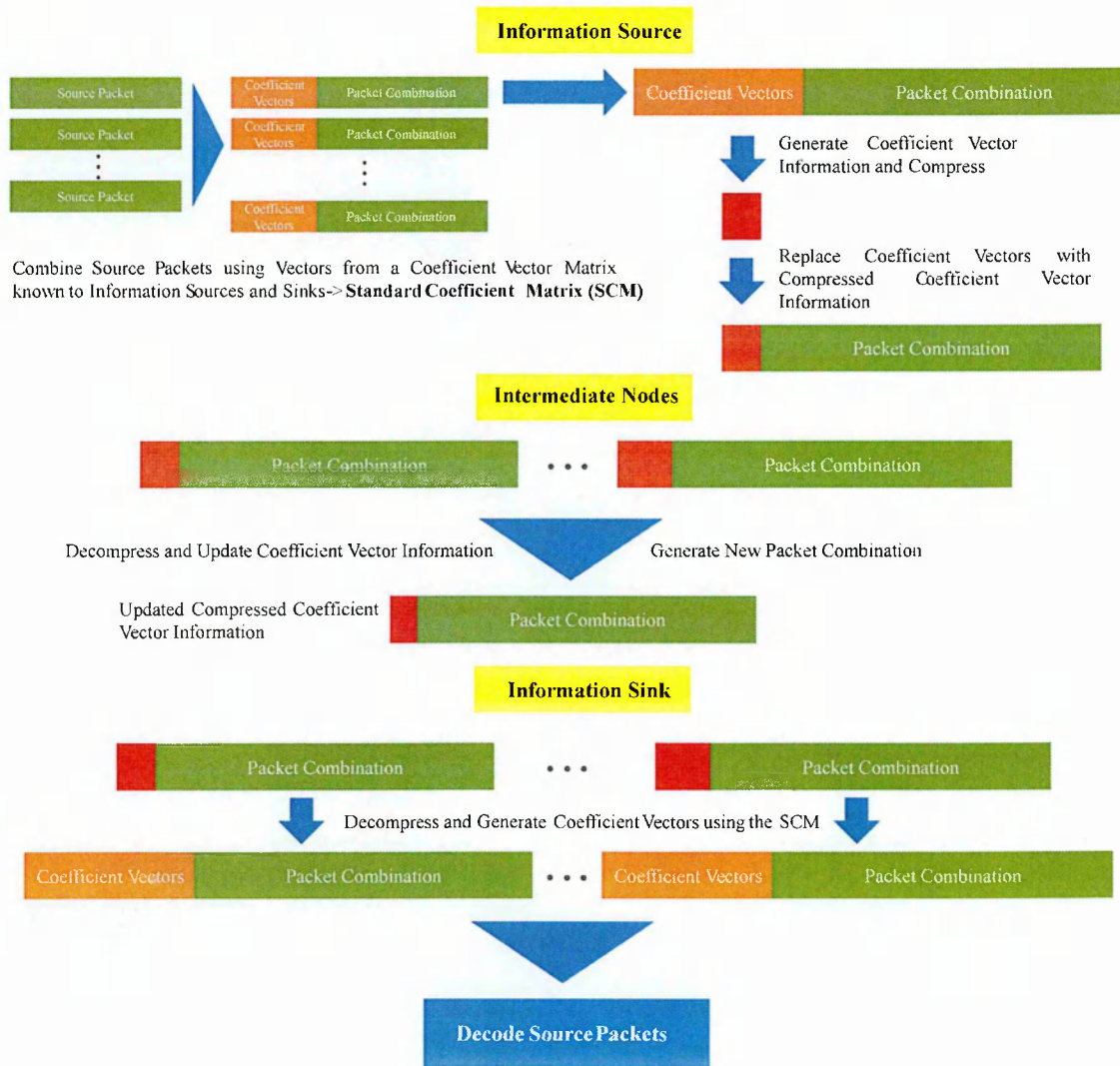
Figure 3-1 Overview of the proposed algorithm

3.2.1 System Model

3.2.1.1 Source Node

The operations performed at the source node to compress PNC coefficients are summarized in Figure 3-2. Accordingly, the source node selects n number of packets $(x_1, x_2, \ldots x_n)$, where, each packet is a $1 \times s$ vector of elements from a Galois field 2^F. F is the field size. These n number of packets belong to the generation g. Using an $m \times n$ coefficient matrix ($m \geq n$), where, each coefficient is also a non-zero element from the same Galois field 2^F, the source combines n packets to generate m linear combinations $(y_1, y_2, \ldots y_m)$.

$$\begin{bmatrix} y_1 \\ y_2 \\ \vdots \\ y_m \end{bmatrix}_{m \times 1} = \begin{bmatrix} c_{1,1} & c_{1,2} & \cdots & c_{1,n} \\ c_{2,1} & c_{2,1} & \cdots & c_{2,n} \\ \vdots & \vdots & & \vdots \\ c_{m,1} & c_{m,2} & \cdots & c_{m.n} \end{bmatrix}_{m \times n} \begin{bmatrix} x_1 \\ x_2 \\ \vdots \\ x_n \end{bmatrix}_{n \times 1} \tag{3-1}$$

The $m - n$ number of redundant packet combinations are transmitted per each generation to increase the robustness of PNC against packet losses, as discussed in Section 2.3. The proposed method considers a coefficient matrix, which is the $m \times n$ portion of a larger matrix, identified as the Standard Coefficient Matrix (SCM). This matrix is known to all source nodes and sink nodes in the network. The SCM consists of non-zero elements from the Galois field 2^F and is generated in such a way that the rank of the $n \times n$ portion of the matrix is n. Therefore, each packet is a linear combination of all the packets in the generation. This increases the probability of a packet combination being useful for multiple sinks in a network with packet losses.

In conventional PNC, the coefficients that are used to generate the particular combination will also be sent along with the packet. Hence the output packet y_v, where $1 \leq v \leq m$, is in the form of,

$$[c_{v,1} \quad c_{v,2} \quad \ldots \quad c_{v,n} \quad | \quad y_v]$$
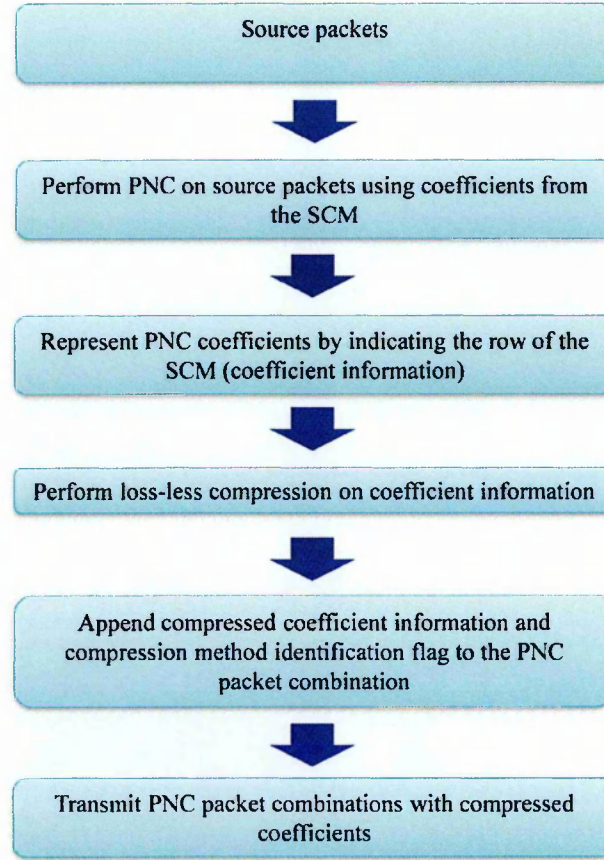
Figure 3-2 Block diagram of source node operations

In contrast, the proposed algorithm indicates which row/rows of the coefficient matrix are used to generate a particular packet combination. This information is identified as coefficient information. Each element of the coefficient information, i_{v_w}, where $1 \leq v \leq m$ and $1 \leq w \leq m$, is represented by j number of bits (calculation of j is later discussed in Section 3.2.1.4). Accordingly, the output packet y_v along with its coefficient information, $[i_{v_1} \quad i_{v_2} \quad \cdots \quad i_{v_m}]$, can be represented as,

$$\begin{bmatrix} i_{v_1} & i_{v_2} & \cdots & i_{v_m} & | & y_v \end{bmatrix}$$

Considering that the packet combination y_v is generated using coefficients belonging only to the vth row of the coefficient matrix, elements of coefficient information for y_v are,

$$i_{v_w} = \begin{cases} 1 & ; v = w \\ 0 & ; v \neq w \end{cases} \tag{3-2}$$

Furthermore, since not all source packet combinations from a generation get combined in a network [8], it is understood that majority of the elements in the coefficient information remains to be zero. Leveraging this observation, coefficient information are compressed using a lossless compression algorithm to minimize the coefficient vector overhead. A simple compression algorithm that can be used is to indicate the position of non-zero elements and their values. The position of a non-zero element can be indicated using $\lceil log_2(generation\ size) \rceil$ number of bits and the i_{v_w} value (representing $(i_{v_w}\ value -$ 1) is sufficient since only non-zero elements are considered) can be represented by j number of bits. However, depending on the coefficient information, a more suitable lossless compression algorithm can be selected out of several suitable algorithms, such as Huffman coding, Golomb coding, arithmetic coding, etc. if necessary. The selected compression algorithm can be indicated using e number of bits from the packet header. Value of e depends on the number of compression algorithms considered by network nodes. These e bits are identified as the compression method identification flag. However, if compression requires sending more bits than it would require sending uncompressed coefficient information, the smaller of either the compressed or uncompressed coefficient information can be sent. In such a case, the compression method identification flag is set to null.

3.2.1.2 Intermediate Nodes

Figure 3-3 illustrates the operations performed at an intermediate node. When an intermediate node receives a single combination from a generation, it will simply forward it to the next hops. If a node receives multiple combinations from the same generation, it adds the respective elements of the linear combinations over the considered finite field.
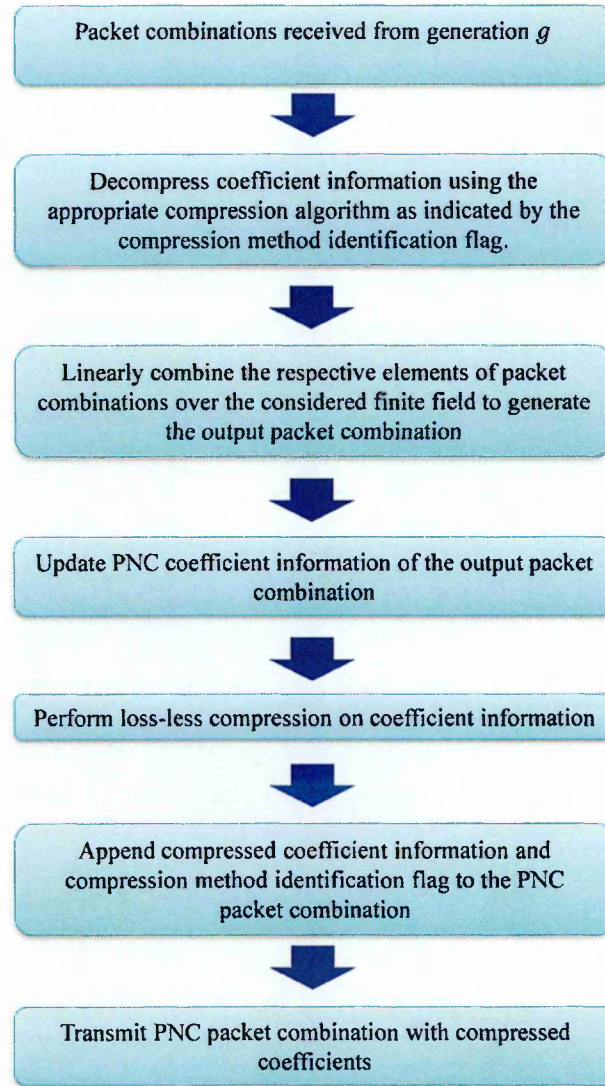
Figure 3-3 Block diagram of intermediate node operations

If an intermediate node receives l combinations $(y_1', y_2', \dots y_l')$, the output combination y'' is generated as,

$$y'' = [1 \quad 1 \quad \cdots \quad 1]_{1 \times l} \begin{bmatrix} y_1' \\ y_2' \\ \vdots \\ y_l' \end{bmatrix}_{l \times 1} \tag{3-3}$$

Using the compression method identification flag, the particular algorithm used to compress the coefficient information can be identified and the compressed coefficient information is decompressed using the appropriate decompression algorithm. Then the coefficient information $(i''_1 \quad i''_2 \quad \cdots \quad i''_l)$ is updated by separately adding the m number of k bits belonging to each combination together. The updated coefficients are,

$$[i''_1 \quad i''_2 \quad \cdots \quad i''_m]_{1 \times m}$$

$$= [1 \quad 1 \quad \cdots \quad 1]_{1 \times l} \begin{bmatrix} i'_{11} & i'_{12} & \cdots & i'_{1m} \\ i'_{21} & i'_{22} & \cdots & i'_{2m} \\ \vdots & \vdots & & \vdots \\ i'_{l1} & i'_{l2} & \cdots & i'_{lm} \end{bmatrix}_{l \times m} \tag{3-4}$$

Hence the output NC packet combination with uncompressed coefficient information is in the form of,

$$\begin{bmatrix} i''_1 & i''_2 & \cdots & i''_m & | & y''_1 \end{bmatrix}$$

Similar to the source node, intermediate nodes select the best algorithm to compress the coefficient information. The compression method identification flag is updated accordingly. Then the compressed coefficient information along with the linear packet combination is transmitted to the next hops.

3.2.1.3 Sink Nodes

The sink node operations are summarized in Figure 3-4. Accordingly, a sink node decompresses the compressed coefficient information and decodes the source packets after receiving sufficient NC packet combinations. Using coefficient information, the sink can identify, which of the coefficient vectors in the coefficient matrix have been used
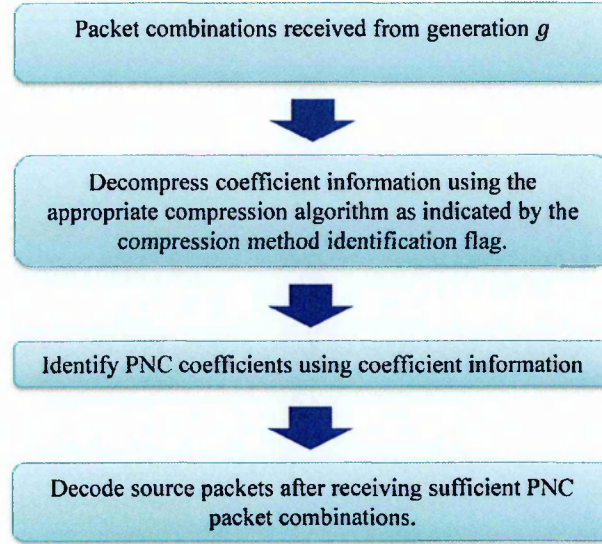
Figure 3-4 Block diagram of sink node operations

in generating each combination as the SCM is readily available in the sink. If the decompressed coefficient information of a packet received by an information sink is,

$$[\hat{i}_1 \quad \hat{i}_2 \quad \cdots \quad \hat{i}_m]$$

the coefficient vector of the received packet is,

$$[\hat{c}_1 \quad \hat{c}_2 \quad \cdots \quad \hat{c}_n]_{1 \times n}$$

$$= [\hat{i}_1 \quad \hat{i}_2 \quad \cdots \quad \hat{i}_m]_{1 \times m} \begin{bmatrix} c_{1,1} & c_{1,2} & \cdots & c_{1,n} \\ c_{2,1} & c_{2,1} & \cdots & c_{2,n} \\ \vdots & \vdots & & \vdots \\ c_{m,1} & c_{m,2} & \cdots & c_{m,n} \end{bmatrix}_{m \times n} \quad (3\text{-}5)$$

The sink collects n packet combinations $(\hat{y}_1, \hat{y}_2, \dots \hat{y}_n)$, with linearly independent coefficient vectors. Then the original packets are calculated as,

$$
\begin{bmatrix} \hat{x}_1 \\ \hat{x}_2 \\ \vdots \\ \hat{x}_n \end{bmatrix}_{n \times 1} = \begin{bmatrix} \hat{c}_{1,1} & \hat{c}_{1,2} & \cdots & \hat{c}_{1,n} \\ \hat{c}_{2,1} & \hat{c}_{2,1} & \cdots & \hat{c}_{2,n} \\ \vdots & \vdots & & \vdots \\ \hat{c}_{n,1} & \hat{c}_{n,2} & \cdots & \hat{c}_{n.n} \end{bmatrix}_{n \times n}^{-1} \cdot \begin{bmatrix} \hat{y}_1 \\ \hat{y}_2 \\ \vdots \\ \hat{y}_n \end{bmatrix}_{n \times 1}
\tag{3-6}
$$

If no packet errors occur, the calculated information will be equal to the original packets.

3.2.1.4 Calculating the Number of Bits Required to Represent Coefficient Information

The number of bits required to represent a coefficient vector using the proposed method, j should be carefully selected ensuring no source packet is combined more than 2^j times. This is performed by initially transmitting probe NC packet generations. Each NC packet generation consists of m NC packets. The number of bits allocated for j is gradually increased per each generation such that NC packet combinations belonging to the gth generation uses $j = g$ number of bits to indicate an element of uncompressed coefficient information. In case a source packet will be combined more than 2^j times at an intermediate node, the source will be informed of the generation the particular packet combination belongs to. The source node uses the minimum number of bits for j such that none of the intermediate nodes encounter a particular source packet being combined more than 2^j times.

3.2.2 Simulation Results and Discussion

To analyze the gain achieved by the proposed algorithm in compressing coefficient vectors, simulations are performed on a wired mesh network generated using the Waxman random network topology generator [100], where 100 network nodes are randomly placed in an area of 100×100 square units. The topology of the mesh network is illustrated in Figure 3-5. Accordingly, the source node, which is located at the origin of the graph is required to multicast data to 15 sink nodes that are connected to the source node through a set of intermediate nodes and network links.
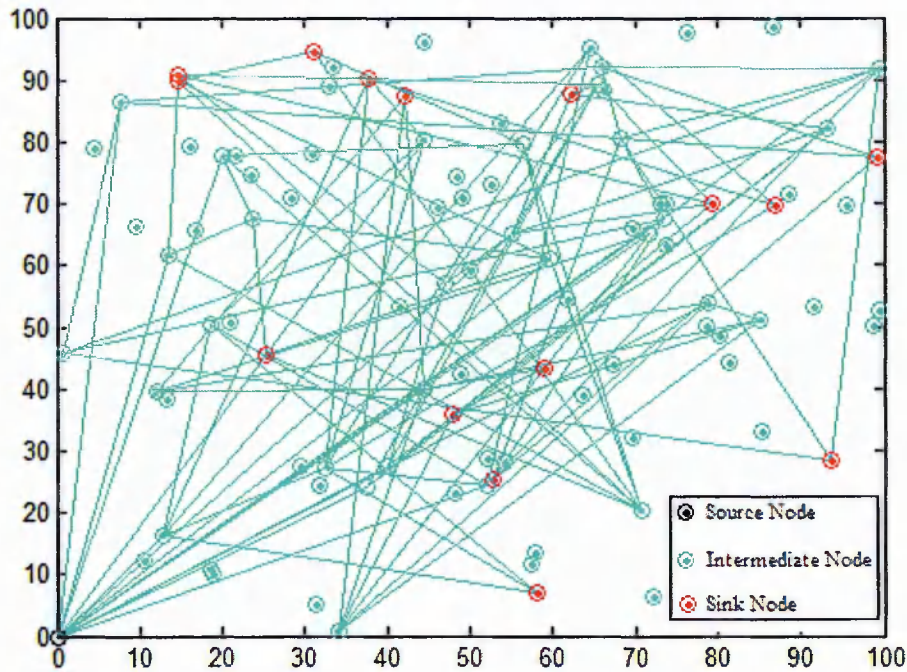
Figure 3-5 Network topology

For the convenience of analysis, header size is assumed to be similar to the number of bytes required to transmit either coefficient vectors or coefficient information. Figure 3-6 illustrates the header sizes in bytes of the proposed algorithm, the compressed coding vector approach proposed in [8] and classical PNC, for a Galois field size of 2^{16}. It should be noted that simulations prove that the proposed algorithm does not affect the decoding rate of NC packets compared to the reference schemes. This lays a fair ground to compare the required header size of the proposed algorithm with competing algorithms.

Analyzing Figure 3-6, it is observed that the header size in classical PNC increases linearly with the generation size. The upper and lower bounds of the header size of the proposed algorithm also increases with the generation size in steps. This occurs due to the requirement of additional bits to indicate the position of non-zero elements while compressing coefficient information. However, the rate at which the upper and lower bounds of the proposed algorithm increase is significantly low compared to both PNC and the compressed coefficient vectors approaches. Therefore, the header size of the proposed algorithm remains significantly smaller compared to other two approaches. Hence, the
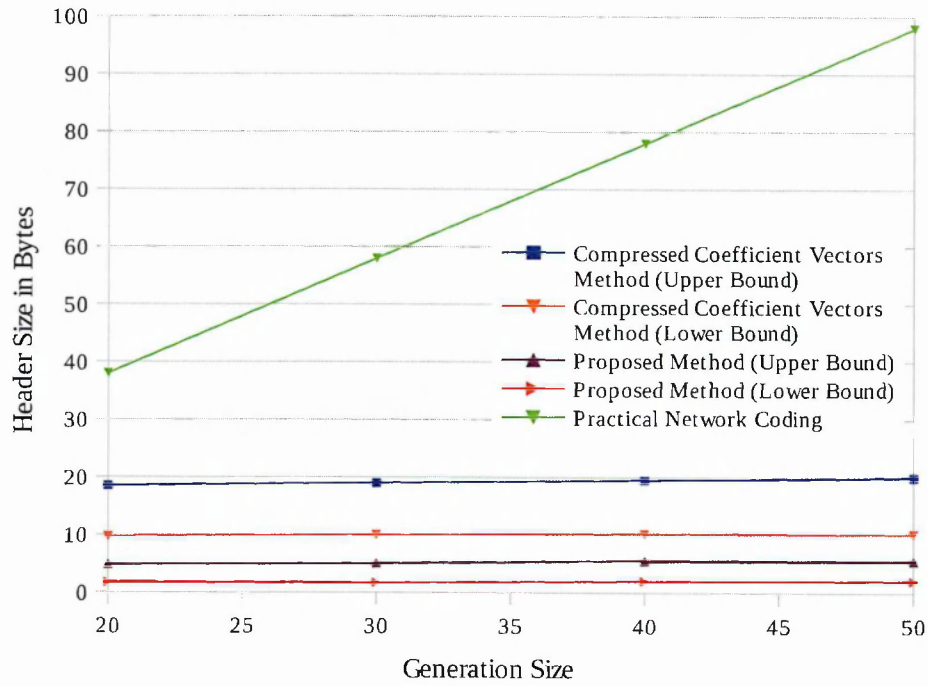
Figure 3-6 Header sizes vs. generation size when the Galois field size is 2^{16}

number of bits per packet required to convey the coefficient vectors to the sink nodes can be minimized. Thus, the $\frac{information\ bytes}{total\ bytes}$ ratio in a packet is increased as seen in Figure 3-7, which in return increases the network throughput.

In addition, it can also be observed that the header size of the proposed algorithm is not dependent on the considered Galois field size, unlike the two reference algorithms. This is due to the number of bits required to transmit the coefficient information being dependent only on the generation size and the network topology. Furthermore, in contrary to the compressed coefficient vectors method, which limits the number of packets that are combined at the source node, the proposed method generates NC packet generations by linearly combining all of the n source packets. This increases the probability of a redundant combination being useful for multiple sinks in a network in case of packet losses.

In the proposed method, network nodes are required to perform compression and decompression operations. Therefore, compression algorithms should be selected considering the computational capabilities of network nodes. However, as networks

evolve to software defined networks, the usage of complex compression algorithms with larger compression gains will be possible. Information sources and sinks are required to know the SCM for each of the considered Galois fields. Even a large $1000x1000$ coefficient matrix, which has elements from a Galois field of size 2^{16} will be just 1.9MB. Therefore, storage of SCM is not a challenge.
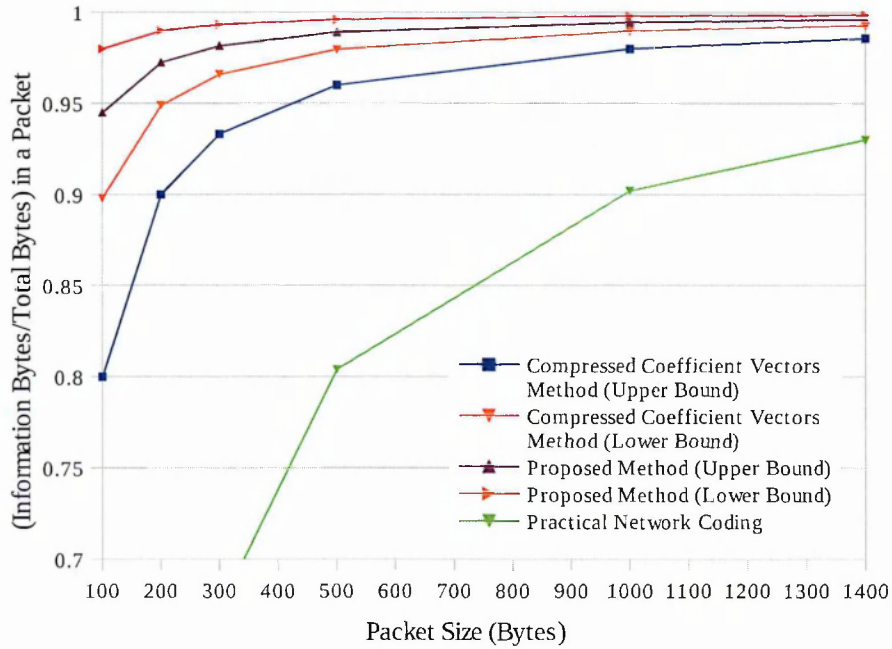
Figure 3-7 Network code rate vs. packet size for a generation size of 50

Furthermore, it should be noted that intermediate nodes are not required to possess knowledge about the SCM. Thus, intermediate nodes are not able to decode source information. Hence, the usage of the proposed method can also be extended to increase data security. Furthermore, as coefficient information includes information on how source packets are combined throughout the network, collecting coefficient information from different paths can be used for network tomography.

In conclusion, it is observed that the proposed method is a feasible solution to minimize the coefficient vector overhead of PNC, which in return increases the throughput of NC enabled networks.

3.3 Delay Sensitive Data Multicast Using Practical Network Coding

Section 2.3.2 of this thesis briefs state-of-the-art feedback schemes available for NC enabled networks. According to Section 2.3.2, it is observed that existing algorithms depend on receiving back to back acknowledgements from sink nodes. Therefore, such schemes are not optimal for delay sensitive data multicast applications in large multi-hop networks, as the delays incorporated can be significant. Furthermore, existing solutions are not capable of transmitting only the appropriate amount of redundancies after sensing the end-to-end packet loss rate. This introduces more than adequate redundancies to a network, which in return increase network congestion. Therefore, a viable solution for NC based delay sensitive multicast applications is to use FEC, by adding the appropriate amount of redundant coded packets. However, determining the required number of redundant coded packets requires knowledge of end-to-end packet erasures. Yet, due to each node transmitting newly coded packets, an end-to-end packet loss rate cannot be defined for NC enabled networks [10].

In response, this section proposes a method to estimate the probability of an NC packet reaching a sink node. This estimate is the used to evaluate the probability of a sink node decoding an NC generation of size n, when the source transmits $m (\geq n)$ coded packets. Using this method, a source can calculate the exact number of coded packets needed to be transmitted per NC generation to deliver delay sensitive data at the a.) required delivery probability b.) optimal throughput. The proposed system overview is presented in Figure 3-8.

3.3.1 System Model

Consider a PNC enabled network, as explained in Section 2.3. The source node selects n packets to generate $m (\geq n)$ linear packet combinations using random coefficients selected from a Galois Field of size 2^F. An $F \geq 8$ value is selected, which is sufficiently large for PNC enabled networks [65]. Each packet is a set of s symbols from the considered Galois field. The coded packets, along with their coding coefficients are transmitted across a packet lossy network. $(m - n)$ redundant coded packets are injected to increase the probability of a sink node decoding an NC generation, P_{dec}. Intermediate nodes further code the received coded packets from the same generation and transmit newly coded packet combinations along with modified coefficients. Upon receiving NC
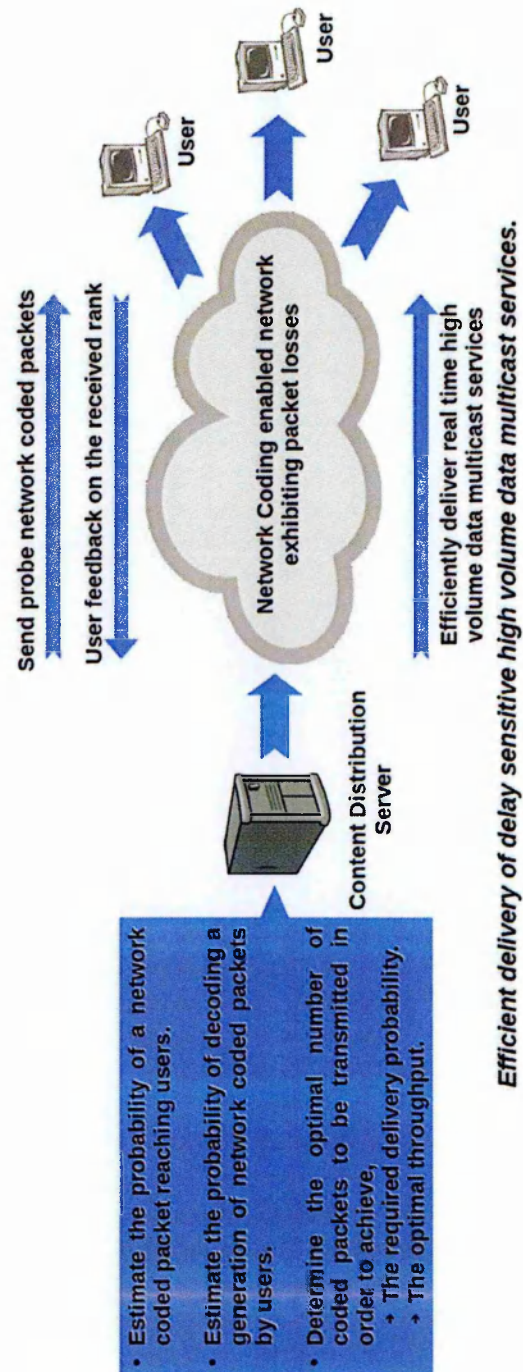
Figure 3-8 System overview for the proposed approach of delay sensitive data multicast using practical network coding

packets such that the rank of coefficients is n, sinks recover the original data. If the rank of the coefficients of received NC packets are less than n, sinks are unable to recover the entire block of n source data packets.

The source node can increase P_{dec} by increasing m and thereby increasing the number of redundant coded packets, $(m - n)$. A larger m decreases the efficiency of bandwidth resource utilization while a lower m reduces P_{dec}. Hence, the optimal value for m should be estimated to deliver packets in a bandwidth efficient manner without compromising P_{dec}.

3.3.1.1 Estimating the Probability of Decoding an NC Generation

The source periodically transmits t number of probe linear packet combinations generated using coefficients with rank t to φ number of sink nodes. Each sink node informs the source of the received number of innovative probe coded packets (rank of received packets). Let the probability of a coded packet reaching a user be α. α is calculated as,

$$\alpha = \frac{rank\ of\ received\ packets}{t} \tag{3-7}$$

Let the probability of a coded packet reaching each of the φ number of users be $\alpha_1 ... \alpha_\varphi$. Then, the probability of a coded packet reaching an average user, α_{avg}, can be calculated as,

$$\alpha_{avg} = \frac{\sum_{v=1}^{\varphi} \alpha_v}{\varphi} \tag{3-8}$$

Hence, the probability of an average user receiving at least n number of coded packets from m number of transmitted coded packets, P_{dec} can be calculated using the binomial distribution as,

$$P_{dec}(n, m, \alpha_{avg}) = \sum_{v=n}^{m} \binom{m}{v} \alpha_{avg}^{v} (1 - \alpha_{avg})^{m-v} \tag{3-9}$$

A value for t is selected empirically such that a consistent value could be obtained for P_{dec}. However, it should be noted that using a larger value for t will increase the accuracy of estimating P_{dec} at the cost of increased decoding complexity, delay and bandwidth usage.

Using (3-8) and (3-9), an information source can calculate the number of coded packets to be transmitted per generation to achieve the required $P_{dec}(n, m, \alpha_{avg})$. Furthermore, (3-8) and (3-9) can be used to find the optimal value for m at which the network throughput is maximized, as explained in Section 3.3.1.2.

3.3.1.2 Maximizing the Network Throughput

Each of the m coded packets with s data symbols contains n coefficients from the Galois Field of size 2^F indicating how it is coded. Thus, the network code rate, ρ_{nw} is,

$$\rho_{nw} = \frac{n}{m} \cdot \frac{s}{(s + n)} \tag{3-10}$$

The generation size is assumed to be greater than the min-cut of the network, which is the case in most practical networks. The normalized throughput, $\tau_{norm} = \frac{decoded\ data\ symbols}{total\ transmitted\ symbols}$ at the sink can be presented as,

$$\tau_{norm} = P_{dec}(n, m, \alpha_{avg}) \cdot \rho_{nw} = \left[\sum_{v=n}^{m} \binom{m}{v} \alpha_{avg}^{v} (1 - \alpha_{avg})^{m-v} \right] \cdot \left[\frac{n}{m} \cdot \frac{s}{(s + n)} \right] \tag{3-11}$$

Using (3-11) and a mathematical optimization technique such as hill climbing, m is calculated for a given n and P_{rec} such that τ_{norm} is maximized ($\tau_{norm_{max}}$).

3.3.2 Simulation Results and Discussion

3.3.2.1 Generating Packet Loss Patterns for Simulations

Link packet loss patterns considered for this simulation are generated using the Gilbert Elliot model, which is a widely used channel model to simulate burst error patterns in transmission channels. This model is based on a Markov chain with two states GOOD (or gap) and BAD (or burst), as illustrated in Figure 3-9.
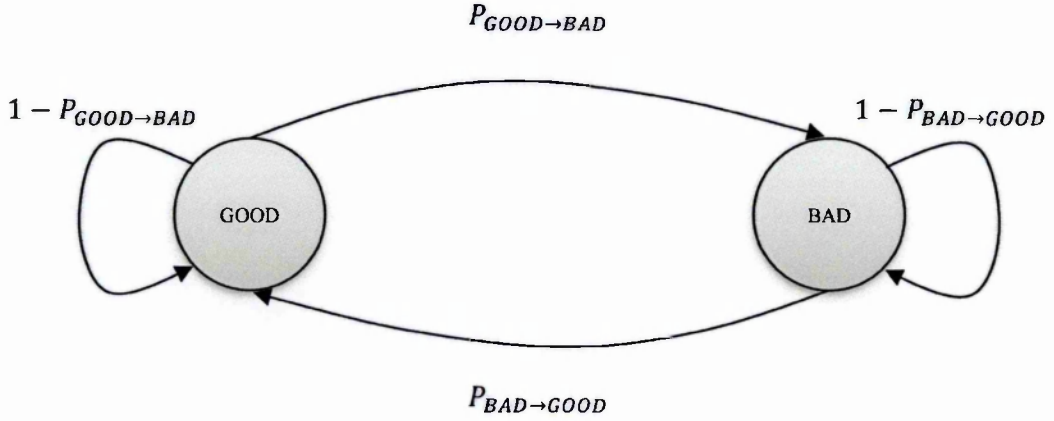
Figure 3-9 The Gilbert Elliot model

The probability of a packet loss in states GOOD and BAD are $P_{LOSS_{GOOD}}$ and $P_{LOSS_{BAD}}$ respectively. Then, the total packet loss rate is defined as,

$$packet\ loss\ rate =$$

$$P_{LOSS_{GOOD}} \cdot \frac{P_{BAD \to GOOD}}{P_{GOOD \to GOOD} + P_{BAD \to GOOD}} + P_{LOSS_{BAD}} \cdot \frac{P_{GOOD \to BAD}}{P_{GOOD \to GOOD} + P_{BAD \to GOOD}} \tag{3-12}$$

It is commonly assumed that no packet losses occur in state G and all packets are lost in state B. Hence, by substituting $P_{LOSS_{GOOD}} = 0$ and $P_{LOSS_{BAD}} = 1$ for (3-12), the packet loss rate can be simplified as,

$$packet\ loss\ rate = \frac{P_{GOOD \to BAD}}{P_{GOOD \to GOOD} + P_{BAD \to GOOD}} \qquad (3\text{-}13)$$

Then, the average length of a packet loss burst is presented as [101], [102],

$$average\ length\ of\ a\ packet\ loss\ burst = \frac{1}{P_{BAD \to GOOD}} \qquad (3\text{-}14)$$

Using (3-13) and (3-14), link packet loss patterns are generated for a considered packet loss rate at a given average packet loss burst length.

3.3.2.2 Simulation Setup

The same network topology described in Section 3.2.2 is simulated to verify the accuracy of the proposed method to estimate the probability of decoding an NC generation. A Galois Field size of 2^8, which is sufficiently large for practical applications [65] is considered. Each coded packet contains $s = 1400$ data symbols.

Figure 3-10 present α_{avg} against the number of probe packets transmitted for two scenarios where α_{avg} converges to 0.98 and 0.95 respectively. According to Figure 3-10, a consistent α_{avg} value for the simulated network could be obtained by multicasting $t = 1000$ probe packets. Hence, this value is used for subsequent simulations.

3.3.2.3 Simulation Results

The α_{avg} value obtained using by multicasting $t = 1000$ probe packets is applied to (3-9) in order to calculate the estimated probability of decoding an NC generation at the particular node. Next, a 1.35MB file is repeatedly multicast over 1000 times to all users in the simulated network. Subsequently, the probability of decoding an NC generation obtained through simulations is compared against estimated values. Figure 3-11 illustrates the estimated vs. simulated values of the probability of decoding a generation for a selected user in the simulated multicast network at different network conditions and different redundancy levels. According to the results presented in Figure 3-11, it can be observed that the estimated probability of decoding an NC generation closely correlates to the simulated results.
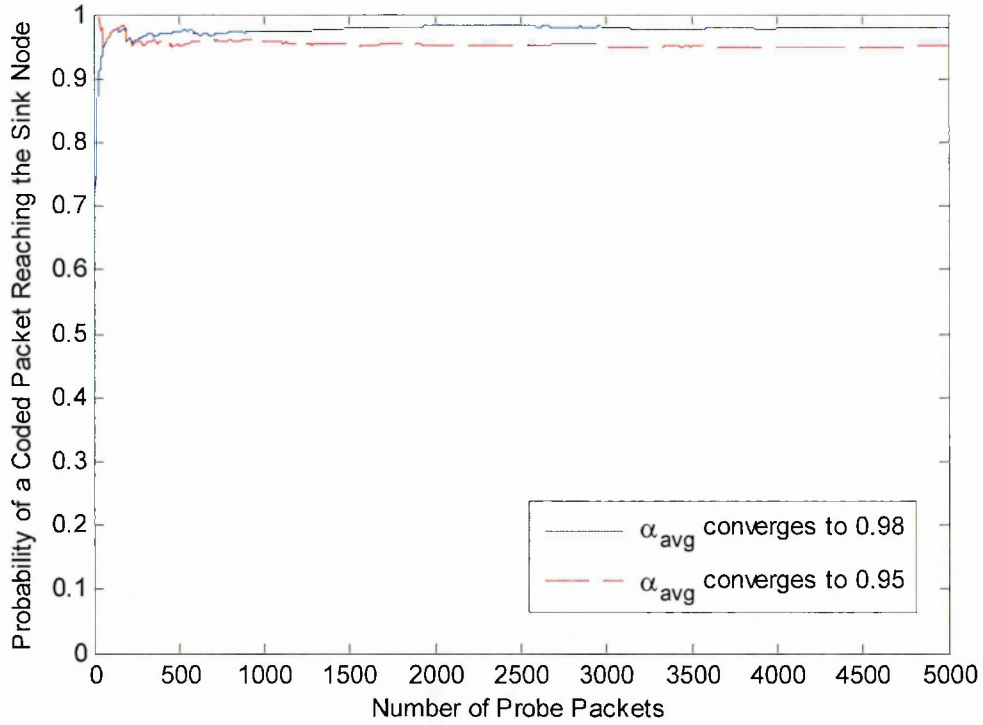
Figure 3-10 Probability of a coded packet reaching a sink node (α_{avg}) vs. the number of probe packets transmitted when α_{avg} converges to 0.98 and 0.95

The correlations coefficient between the estimated and simulated probabilities is over 0.98. This validates the proposed method to estimate the probability of an NC packet reaching a user. Hence, the proposed method can be used to calculate the required number of redundant packets per generation to be transmitted such that NC packets can be delivered at the required packet decoding probability.

Next, using (3-11), the optimal number of coded packets required to be transmitted, m at $n = 50$ to achieve $\tau_{norm_{max}}$ is calculated for different values of α_{avg}. Figure 3-12 presents the maximum throughput obtained using the proposed method, compared with state-of-the-art ARQ based feedback methods, namely Traditional Feedback and Freedom Feedback, explained in [25]. Even with the advantage of not depending on back to back ARQ, the proposed method exhibits an increased end-to-end throughput. This is due to the proposed method estimating the network accurately and transmitting only the required amount of coded packets per generation using (3-11), thereby less congesting the network compared to reference methods. Hence, the proposed method can be used to multicast delay sensitive data, achieving the optimal throughput.
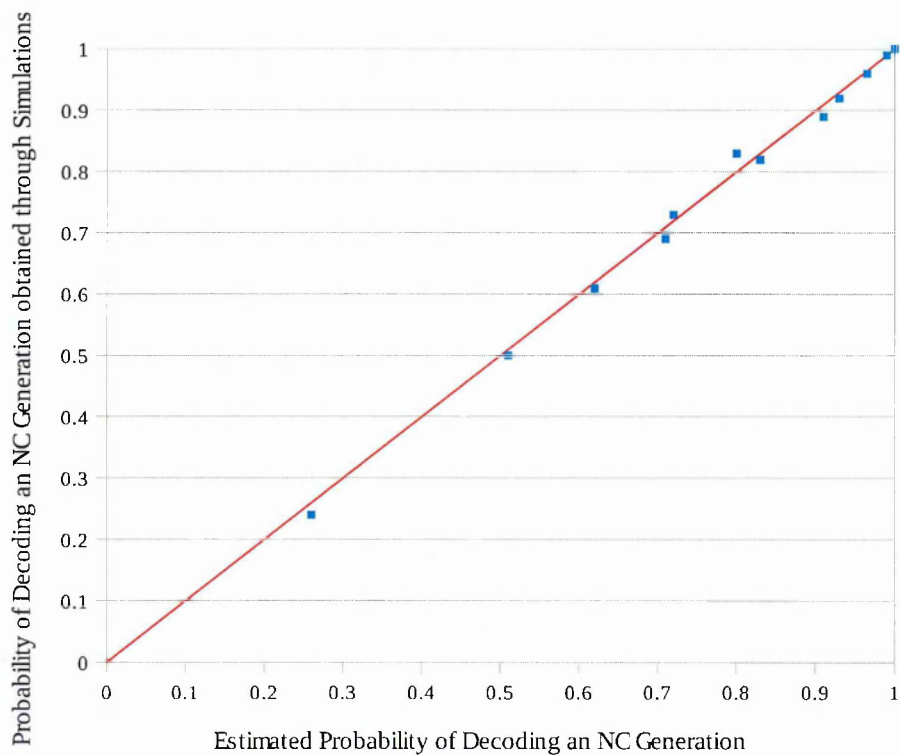
Figure 3-11 Probability of decoding an NC generation obtained through simulations vs. estimated probability of decoding an NC Generation
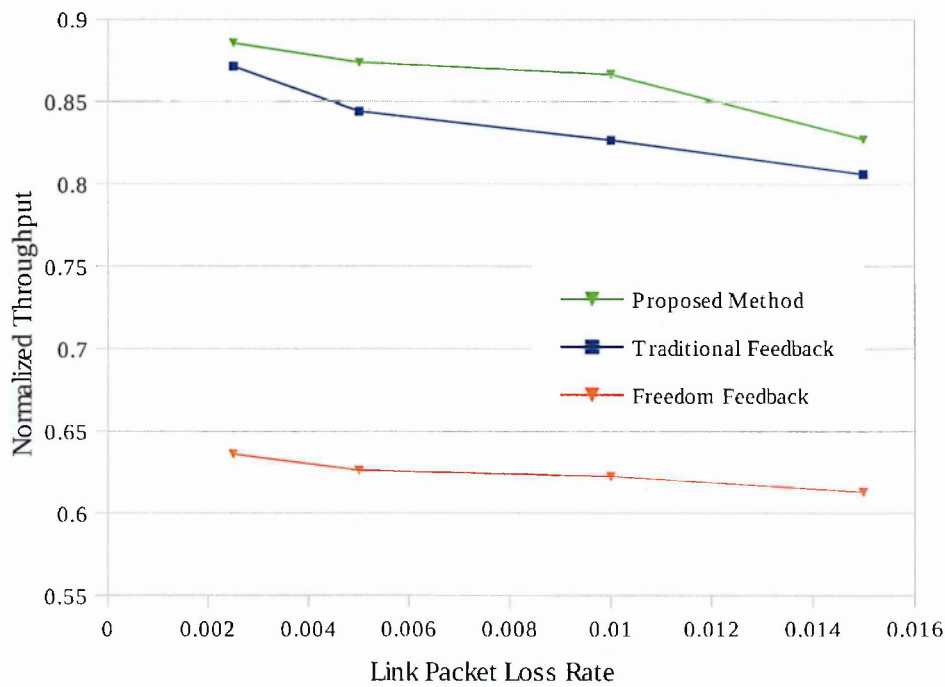
Figure 3-12 Simulation results of normalized throughput vs. link packet loss rate

3.4 Practical Network Coding with Pre-Coding and Interleaving

PNC introduces the concept of performing RLNC for groups of packets called generations to further extend the usability of network coding in realistic networks with packet erasures, delays and topology changes. In addition, PNC allows transmitting redundant network coded packets as a Forward Error Correction mechanism to increase the probability of decoding NC generations amidst packet erasures. These redundant NC packets perform FEC more efficiently than traditional ARQ based schemes to combat packet erasures [21]. However, it should be noted that packet losses in the internet is bursty and only a relatively small number of packet loss bursts account for most of the packet losses [103]. When packet losses occur in bursts, a significant amount of packet combinations from a generation can be lost such that the redundant NC packets in the generation are not sufficient to decode an NC generation. In response, this chapter proposes the novel approach of Practical Network Coding with Pre-Coding and Interleaving to increase the resilience of PNC against packet losses.

3.4.1 System Model

PNCPCI incorporates interleaving and pre-coding to PNC in order to recover lost packet combinations efficiently and increase the robustness against packet losses. Random packet losses are recovered during decoding PNC combinations while the burst packet losses which are spread after deinterleaving are recovered during decoding pre-coded combinations. Figure 3-13 illustrates the block diagram of the PNCPCI process proposed to be used for robust video communication in a network with packet losses.

3.4.1.1 Packet Encoding at the Source Node

PNCPCI performs pre-coding and interleaving operations at the source to increase the robustness of a packet transmission against burst packet losses. The pre-coding operation linearly combines source packets to generate a set of linear packet combinations, as explained below. Let $[X_M]_{n \times s} (M = 1,2, \dots, m)$, where $m \geq n$, be $n \times m$ source packets where each packet is a $1 \times s$ matrix of symbols from a Galois field of size 2^F. F is the order of the Galois field.
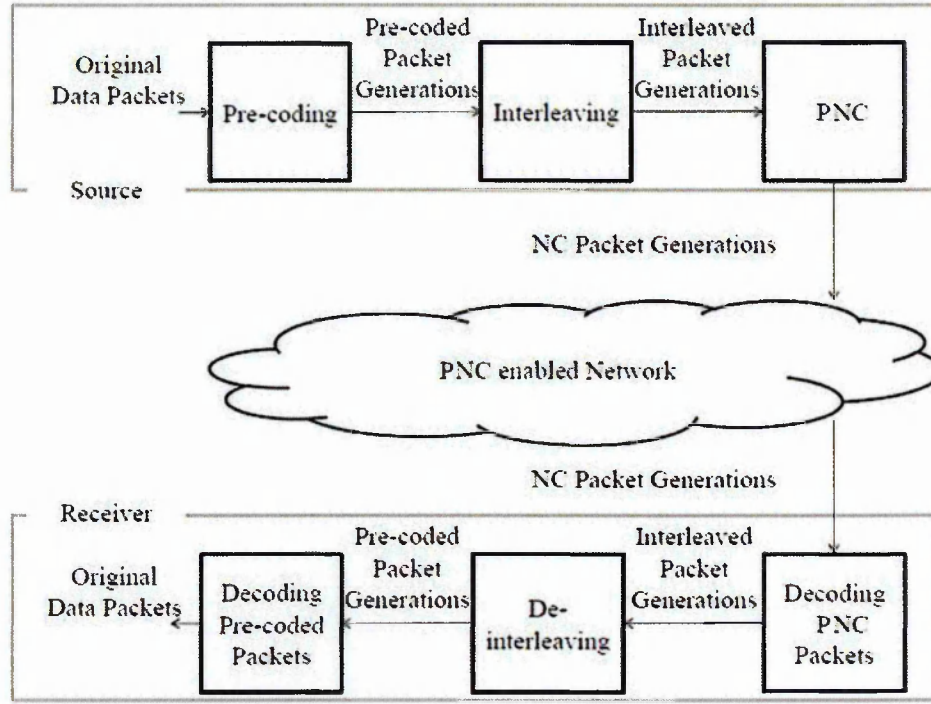
Figure 3-13 Block diagram of Practical Network Coding with Pre-Coding and Interleaving

Pre-coding generates $m \times m$ number of linear combinations, $[Y_M]_{m \times s} (M = 1,2, ..., m)$, by performing,

$$[Y_M]_{m \times s} = \omega_{m \times n}.[X_M]_{n \times s} \qquad (3\text{-}15)$$

ω is a $m \times n$ portion of the same SCM discussed in Section 3.2.1.1, which is known to both the sender and the receiver.

In order to increase the robustness against packet loss bursts, the pre-coded packet combinations are interleaved with each other to generate $[\acute{Y}_\delta]_{m \times p} (\delta = 1,2, ..., m)$ such that,

$$[\acute{Y}_\delta]_{m \times s} = \gamma([Y_M]_{m \times s}) \qquad (3\text{-}16)$$

γ is performed by,

$$\acute{Y}_\delta(M, w) = Y_M(\delta, w) \tag{3-17}$$

where w $(1 \leq w \leq s)$ is the position of a symbol in a packet.

Next, PNC is applied to $\left[\acute{Y}_\delta\right]_{m \times s} (J = 1,2, ..., m)$ to generate $[Z_\delta]_{l \times s}(\delta = 1,2, ..., m)$, where $l \geq m$. NC generations are generated using $[C_\delta]_{l \times m}(\delta = 1,2, ..., m)$, where each element of C_δ is a symbol from a Galois field of size 2^F. The output linear packet combinations, $[Z_\delta]_{l \times s}(\delta = 1,2, ..., m)$ are generated as,

$$[Z_\delta]_{l \times s} = [C_\delta]_{l \times m} \cdot \left[\acute{Y}_\delta\right]_{m \times s} \tag{3-18}$$

These $m \times l$ number of output packet combinations $(Z_1 ... Z_m)$ are identified to be in the same interleaved group. Each set of l output linear packet combinations, represented as $[[C_\delta]_{l \times m}[Z_\delta]_{l \times s}]$, belong to one NC generation. The information related to NC generation, interleaved group and sequence number (position of a particular NC generation in the interleaved group) are also appended to each packet in order to provide the required information for sink nodes to decode PNCPCI coded packets. Subsequently, the linear packet combinations are multicast over a PNC enabled network.

3.4.1.2 Intermediate Nodes

Intermediate nodes perform PNC to linearly combine packets belonging to the same generation. It should be noted that all packets belonging to the same generation possess the same interleaved group and the sequence number. Hence the NC generation, interleaved group and the sequence number are also appended to output packet combinations.

3.4.1.3 Packet Decoding at a Sink Nodes

Let the packets received by the sink node form an interleaved group be in the form of $\left[[\hat{C}_\delta]_{\eta_\delta \times m}[\hat{Z}_\delta]_{\eta_\delta \times s}\right]$. η_δ is the number of linear packet combinations belonging to the δth

NC generation, received by a sink node. $[C_j]_{\eta_\delta \times m} (j = 1,2,...,m)$ is the received coefficient matrix and $[\hat{Z}_j]_{\eta_\delta \times m} (j = 1,2,...,m)$ is the received linear packet combination matrix. Then, the sink node selects m linearly independent combinations from each NC generation as,

$$\left[[\hat{C}_\delta]_{m \times m} [\hat{Z}_\delta]_{m \times s}\right] = \kappa \left([\hat{C}_\delta]_{\eta_\delta \times m} [\hat{Z}_\delta]_{\eta_\delta \times s}\right) \tag{3-19}$$

Where κ selects m linearly independent combinations from the δth NC generation, provided that $\eta_\delta \geq m$ and the receiver has received m linearly independent packet combinations out of the received η_δ linear packet combinations.

The selected linearly independent packet combinations and coefficients from the δth NC generation are used to regenerate an interleaved generation such that,

$$[\hat{Y}_\delta]_{m \times s} = [\hat{C}_\delta]_{m \times m}^{-1} \cdot [\hat{Z}_\delta]_{m \times s} \tag{3-20}$$

m interleaved generations belonging to an interleaved group are then ordered using the sequence number and deinterleaved to generate pre-coded generations.

$$[\hat{Y}_M]_{m \times s} = \bar{\gamma} [\hat{Y}_\delta]_{m \times s} \tag{3-21}$$

$\bar{\gamma}$ is performed such that,

$$\hat{Y}_M(\delta, w) = \hat{Y}_\delta(M, w) \tag{3-22}$$

n linearly independent combinations from each m sets of deinterleaved combinations belonging to a pre-coded generation are selected along with their respective coefficients

from the SCM. Let the selected coefficients and the combinations be $[\omega_M]_{n \times n}[\hat{Y}_M]_{n \times s}$. Then, the original packets are decoded by performing,

$$[\hat{X}_M]_{n \times s} = [\omega_M]_{n \times n}^{-1} \cdot [\hat{Y}_M]_{n \times s}$$

(3-23)

3.4.2 Simulation Results and Discussion

The same wired mesh network described in Section 3.3.2 is simulated to evaluate the performance of the proposed method of PNCPCI. The results are compared against traditional PNC. Bursty link packet losses are generated using the Gilbert Elliot model. PNCPCI is performed by generating half of the redundancies ($m - n$ number of packets) while pre-coding and the other half of the redundancies ($l - m$ number of packets) while performing PNC. In contrast, all of the redundancies are added while performing PNC to compare the results of the proposed method against classical PNC.

Figure 3-14 illustrate the probability of decoding an NC generation against the probability of a coded packet reaching a randomly selected sink node when the average length of a packet loss burst is 20 packets. The considered redundancy rate is 37.5%. Hence, for $n = 50$ source packets, $m - n = 15$ redundant packets and $l - m = 15$ redundant packets are generated while pre-coding and performing PNC respectively. Classical PNC is performed by generating 80 linear packet combinations per 50 source packets. Accordingly, it is observed that the proposed PNCPCI algorithm is more capable of recovering source packets amidst bursty packet losses compared to traditional PNC. This performance gain is achieved by spreading the burst packet losses across NC generations during de-interleaving such that NC generations can be decoded during the process of decoding pre-coded packets. In addition, it can be observed that the proposed PNCPCI algorithm is capable of decoding all source packets. This qualifies the proposed algorithm to be used in networks, which require the accurate end-to-end delivery of coded packets amidst bursty packet loses. Furthermore, it is also observed that the performance of the proposed algorithm deteriorates at high packet loss rates. This is due to the increased probability of not being able to decode a complete interleaved group of NC generations when the number of packet losses is large. However, with more redundancies, the
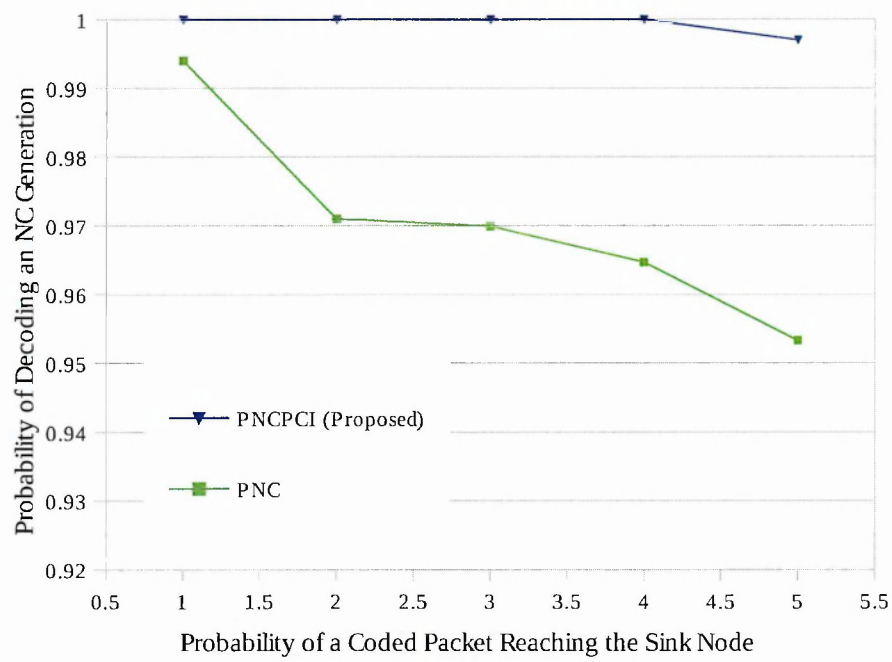
Figure 3-14 Probability of decoding an NC generation vs. the probability of a coded packet reaching the sink node
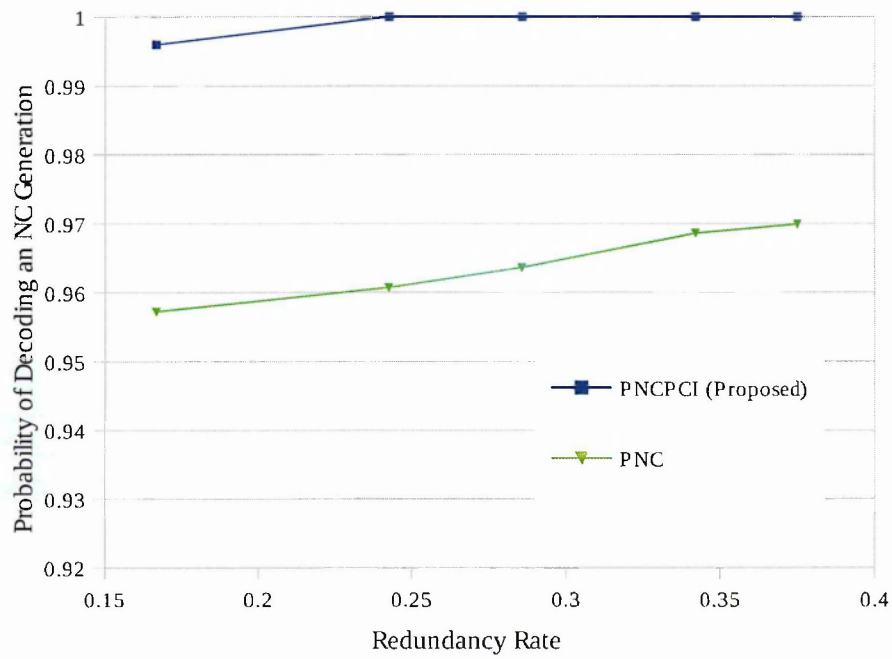
Figure 3-15 Probability of decoding an NC generation vs. the redundancy rate

proposed algorithm is more capable of completely decoding all source packets amidst bursty packet losses, compared to traditional PNC. This is observed in Figure 3-15, which illustrates the probability of decoding an NC generation against the redundancy rate. The results are obtained when the probability of a coded packet reaching the selected sink node is 2% at average packet loss burst lengths 20. In addition, Figure 3-16 illustrates the probability of decoding an NC generation against the average length of a packet loss burst. The results are obtained at a 2% probability of a coded packet reaching the selected sink node and a redundancy rate of 37.5%. These results indicate that the ability of the proposed method over classical PNC in decoding packet loss bursts become is more significant as the length of a packet loss burst increase.

Based on the presented results, it can be concluded that the proposed method of PNCPCI is capable of increasing the robustness of a data multicast against packet loss bursts, compared to classical PNC. Hence the proposed PNCPCI algorithms are capable of increasing the probability of decoding NC generations, which in return increases the throughput of a data multicast in a network under bursty packet losses over existing algorithms.
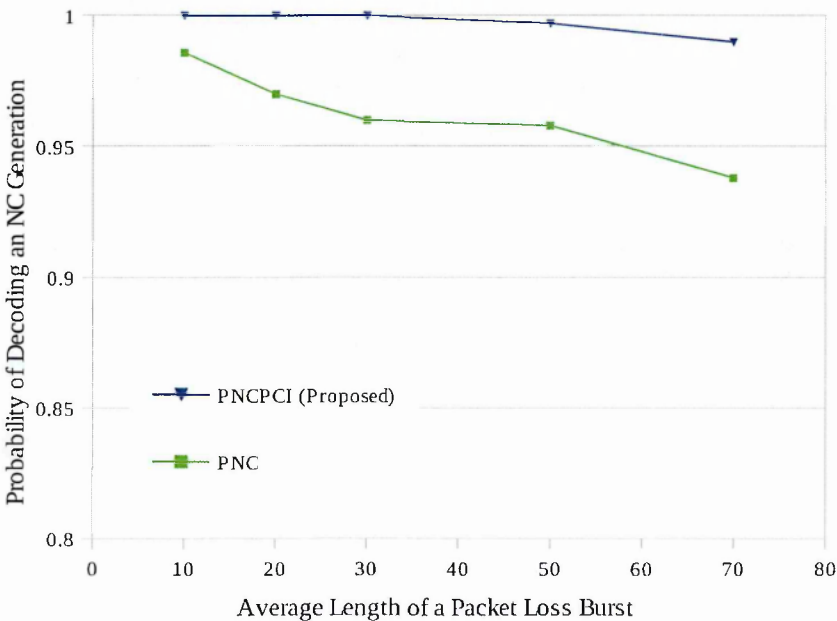
Figure 3-16 Probability of decoding an NC generation vs. average length of a packet loss burst

3.5 Summary

This chapter proposes three novel algorithms of adopting PNC algorithms to be used in realistic network conditions.

An algorithm to minimize the coefficient vector overhead, which in return improves the network throughput is presented in Section 3.2. This is performed by exploiting the fact that the number of source packets that get combined in a PNC network is limited and a lower number of bits compared to state-of-the-art algorithms are sufficient to represent each element of coefficient information. Simulation results indicate that by means of efficiently compressing coefficient vectors, the proposed method is capable of achieving up to 15% improvement in the throughput compared to existing algorithms.

In addition, Section 3.3 proposes a method to estimate the end to end probability of an NC packet being delivered to a sink node. Using the proposed method, the allocation of redundancies is optimized to enhance the FEC capability of PNC. The proposed method does not require a source receiving back to back ARQs. Furthermore, the redundancy allocation for PNC can be optimized to achieve either a predefined packet delivery probability to meet a given QoS requirement or to optimize the throughput to maximize network resource utilization. Simulation results indicate a 0.98 correlation between the simulated and required packet delivery probability. Furthermore, up to 5% gains in throughput is observed using the proposed methods compared to state-of-the-art reference schemes. Thus the proposed method can be used to efficiently deliver real time high volume data multicast services to users.

Furthermore, a novel algorithm, PNCPCI, to improve the robustness of PNC against packet loss bursts is presented in Section 3.4. PNCPCI incorporates interleaving and pre-coding to PNC in order to recover lost packet combinations efficiently and increase the robustness against packet losses. Random packet losses are recovered during the process of decoding PNC combinations while the burst packet losses which are spread after deinterleaving are recovered during decoding pre-coded packet combinations. Simulation results exhibit up to a 4.5% gain in the probability of decoding an NC generation compared to classical PNC. Hence, the proposed method of PNCPCI is capable of increasing the robustness of a data multicast against packet loss bursts, compared to classical PNC. Therefore, the proposed PNCPCI algorithms are capable of increasing the

probability of decoding NC generations over existing algorithms. This in return increases the throughput of a data multicast in a network under bursty packet losses.

Consequently, it is observed that the algorithms proposed Chapter 3 exhibit significant throughput improvements compared to existing PNC algorithms.

Chapter 4

4 Content and Network-Aware Network Coding for Efficient Multicast Applications

4.1 Introduction

This chapter proposes content and network-aware NC algorithms to increase the performance of delay sensitive data and video multicast applications. As pointed out in Section 2.4, NC-MGM improves the robustness of NC against packet losses. However, state-of-the art NC-MGM algorithms are not capable of intelligently allocating redundancies considering the network status and content properties. In response, this chapter proposes algorithms, which optimizes the redundancy allocation for NC-MGM to increase the performance of delay sensitive data and video multicast applications across packet lossy networks. The key contributions presented in this chapter are outlined below.

1. Develop a mathematical model to evaluate the probability of decoding an NC generation in a multi-generation.

2. Using the mathematical model, optimize the redundancy allocation for network coding by considering the network status to increase the probability of decoding NC generations.

3. Consider both the network status and the priority of scalability layers to optimize the redundancy allocation for NC in order to increase the received video quality.

4. Perform a per user optimization to allocate redundancies for NC to increase the probability of continuous video playback at all users in a scalable video multicast.

Rest of this chapter is organized as below. Section 4.2 presents the proposed algorithms. Simulation results of the proposed algorithm are presented in section 4.3. Section 4.4 concludes the chapter.

4.2 System Model

4.2.1 Increasing the Probability of Decoding an NC Generation

This section proposes an optimization algorithm to efficiently allocate redundancies for NC-MGM such that the probability of decoding an NC generation is maximized.

Consider a multicast network consisting of a source node, intermediate nodes and φ number of users. The objective is to transmit the maximum amount of data from the source to users by using available bandwidth resources efficiently. The proposed optimization algorithm is described below. The source node groups data in to sets of $\sum_{d=1}^{z} n_d$ number of source packets where z is the number of generations in an NC-MGM mixing set and n_d is the number of source packets belonging to the generation having the position index d. Each source packet consists of s number of elements selected from a Galois field of size 2^F. The source packets are linearly combined using coefficients selected from the same Galois field to generate $\sum_{d=1}^{z} m_d$ number of NC packets belonging to generations $g_1, g_2, ..., g_z$ in a multi-generation, where m_d is the number of NC coded packets belonging to generation g_d. The m_d number of NC packets from generation g_d, where $0 < d \leq z$, are generated by linearly combining the first $\sum_{v=1}^{d} n_v$ number of source packets. The optimal number of coded packets generated for generation g_d, m_d, to maximize the probability of decoding an NC generation is calculated in the following manner.

The objective of this algorithm is to maximize the probability of decoding NC generations for an average user. Let the probability of a coded packet reaching an average user, α_{avg}. This probability can be calculated using (3-8) presented in Section 3.3.1.1.

It should be noted that a generation g_d and all its preceding generations, $g_1, ..., g_{d-1}$, can be decoded if $\sum_{v=1}^{d} n_v$ number of linearly independent packet combinations are received from generations $g_1, ..., g_d$. Therefore, a receiver node can decode all generations

$g_1, \dots, g_d$ if generation $g_{d'}$ is, where $0 \le d' < d$, is decoded and $\sum\limits_{v=d'+1}^{d} n_v$ number of

linearly independent coded packets are received from the generations $g_{d'+1}, \dots, g_d$. Let

the probability of decoding the generation g_d after receiving packets from only the

generations $g_1, \dots, g_d$ be ς_d. It is assumed that a sufficiently large Galois field size is

considered. Therefore, when $d = 1$, ς_1 is equal to the probability of receiving at least n_1

number of coded packets out of the transmitted m_1 number of coded packets from

generation g_1. Thus, using (3-9),

$$\varsigma_1 = P(n_1, m_1, \alpha_{avg}) \tag{4-1}$$

In similar vein, ς_2 is equal to the probability of receiving at least n_2 packet combinations

from generation g_2 if g_1 is decoded. Else, both generations g_1 and g_2 can be decoded if a

total of $n_1 + n_2$ packet combinations are received from the respective generations.

Therefore, ς_2 is calculated as,

$$\varsigma_2 = \varsigma_1.P(n_2, m_2, \alpha_{avg}) + (1 - \varsigma_1).P\left(\sum_{v=1}^{2} n_v, \sum_{v=1}^{2} m_v, \alpha_{avg}\right) \tag{4-2}$$

Likewise, ς_z can be calculated as,

$$\varsigma_z = \varsigma_{z-1}.P(n_z, m_z, \alpha_{avg}) + (1 - \varsigma_{z-1}).\varsigma_{z-2}.P\left(\sum_{v=z-1}^{z} n_v, \sum_{v=z-1}^{z} m_v, \alpha_{avg}\right)$$
$$+ \cdots \tag{4-3}$$
$$+ (1 - \varsigma_1).(1 - \varsigma_2)\dots(1 - \varsigma_3).P\left(\sum_{v=1}^{z} n_v, \sum_{v=1}^{z} m_v, \alpha_{avg}\right)$$

Considering (4-1), (4-2) and (4-3), ς_d can be presented in a generalized equation as,

$$\varsigma_d = \begin{cases} P(n_1,m_1,\alpha_{avg}) & ;d = 1 \\ \left[\begin{array}{l} \varsigma_{d-1}.P(n_d,m_d,\alpha_{avg}) \\ +\sum_{v=1}^{d-1}\left[\varsigma_{d-v-1}.\prod_{w=d-v}^{d-1}\left((1-\varsigma_k).P(\sum_{w=d-1}^{d}n_w,\sum_{w=d-1}^{d}m_w,\alpha_{avg})\right)\right] \end{array}\right] & ;d \geq 1 \end{cases} \qquad (4\text{-}4)$$

where, $\varsigma_0 = 1$.

Furthermore, it should be noted that the generation g_d can be decoded if $\sum_{v=d}^{\tilde{d}} n_v$ number of linearly independent packets are received from generations $g_d, ..., g_{\tilde{d}}$, where $d \leq \tilde{d} \leq z$. Let the probability of decoding the generation g_d after receiving packet combinations from all the z generations in the multi-generation be ρ_d. When $d = z$,

$$\rho_z = \varsigma_z \qquad (4\text{-}5)$$

Similarly, when $d = z - 1$,

$$\rho_{z-1} = \varsigma_{z-1} + (1 - \varsigma_{z-1}).\varsigma_z \qquad (4\text{-}6)$$

Likewise, when $d = 1$,

$$\rho_1 = \varsigma_1 + (1 - \varsigma_1).\varsigma_2 + \cdots + (1 - \varsigma_1)...(1 - \varsigma_{z-1})\varsigma_z \qquad (4\text{-}7)$$

Considering (4-5), (4-6) and (4-7), ρ_d is presented in a more generalized manner as stated below.

$$\rho_d = \begin{cases} \varsigma_d & ;d=z \\ \varsigma_d + \sum\limits_{v=d+1}^{z}\left[\varsigma_v.\prod\limits_{w=d}^{v-1}(1-\varsigma_w)\right] & ;d<z \end{cases} \tag{4-8}$$

Using (4-8), the average probability of decoding an NC generation, $\bar{\rho}$, can be calculated as,

$$\bar{\rho} = \frac{\sum\limits_{d=1}^{z}\rho_d}{z} \tag{4-9}$$

The optimal values for $m_1, m_2, \ldots, m_z$, which maximizes the probability of decoding NC generations is found when,

$$Max\left(\bar{\rho}\right)$$

Such that,

$$\sum_{d=1}^{z}\left(m_d\right) = R \tag{4-10}$$

where R is the total number of NC packet combinations that can be transmitted per multi-generation within the allocated bandwidth.

4.2.2 Increasing the Received Video Quality

One of the significant challenges in using NC technologies for multimedia communications (as discussed in Section 2.6) is to take the unequal packet importance of video content in to consideration in order to protect high priority video data amidst packet losses. Addressing this, a number of algorithms available in the literature propose to perform NC while generating redundancies such that better protection is applied to high priority video layers in order to enhance the robustness of a scalable video multicast against packet losses (Sections 2.6.1-2.6.2). However, these algorithms are not optimal in

allocating redundancies for PNC to deliver high volume video multicast services efficiently across networks exhibiting packet losses.

As a solution, the optimization algorithm described in Section 4.2.1 for data multicast is optimized to increase the received video quality in a scalable video multicast. This is performed by also taking the priority of scalable video layers in to consideration in order to find the optimal redundancy allocation such that the received video quality of an average user is maximized.

Consider a scalable video with z scalability layers. These scalability layers are generated such that each consecutive video layer is encoded at a higher quality by controlling either the quantization parameter, spatial resolution or temporal resolution. Hence, each consecutive scalability layer received by a user will increase the received video quality. All the video slices belonging to the dth scalability layer in a GOP are assigned to the generation g_d, as illustrated in Figure 4-1. Similar to Section 4.2.1, the optimization is performed considering the average user, with a probability of a coded packet reaching the user being α_{avg}.

The received video quality of an average user is estimated in the following manner. The probability of receiving all scalability video layers only up to the dth scalability layer, σ_i, is equal to the probability of decoding generations $g_1, ..., g_i$ and not decoding the generation g_{i+1}. Hence, σ_i can be calculated as,

$$\sigma_d = \left[\prod_{v=1}^{d} \rho_v \right] \cdot [1 - \rho_{d+1}] \tag{4-11}$$
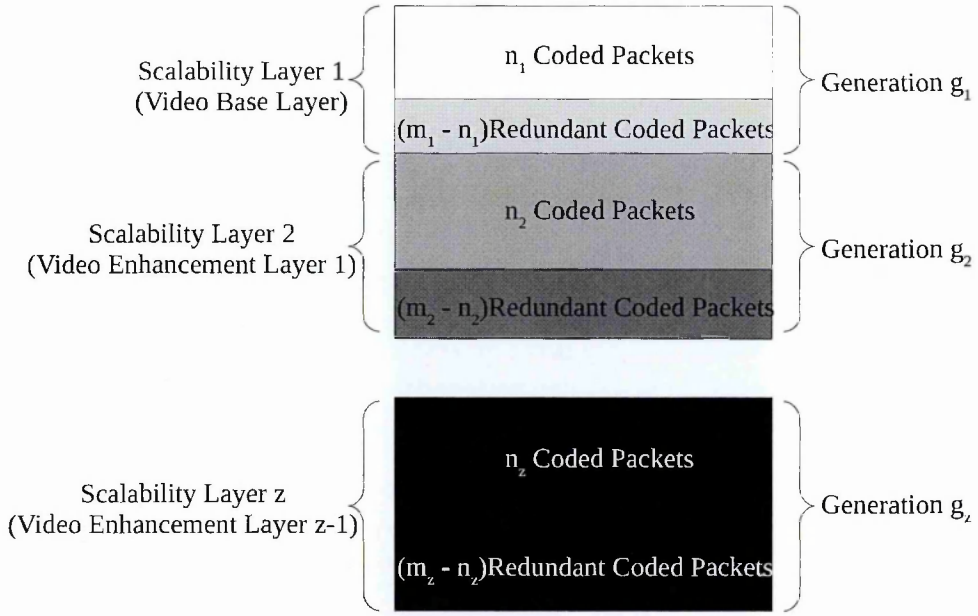
Figure 4-1 Assignment of scalable video packets in to NC generations in a multi-generation

Let the quality of the video when all layers up to the dth scalability layer is received be q_d. Then, the received video quality of the average user, ϕ can be estimated as,

$$\phi = \sum_{d=1}^{z} \sigma_d . q_d \tag{4-12}$$

The optimal $m_1, m_2, ..., m_z$ values, which maximizes the received video quality at the average user is calculated by performing the following optimization.

$$Max(\phi)$$

Such that,

$$\sum_{d=1}^{z} (m_d) = R \tag{4-13}$$

4.2.3 Increasing the Probability of Continuous Video Playback at All Users

The algorithm proposed in Section 4.2.2 allocates redundancies such that the video quality received by an average user is maximized. However, this approach may decrease the probability of at least decoding the base layer of a scalable video for users having below average network conditions. Without decoding the base layer, a user is not able to playback the video continuously. Therefore, increasing the probability of decoding the base layer of a video at all users will in return increase the probability of continuous video playback and thereby increase user satisfaction. Addressing this, an optimization algorithm is proposed in this Section to distribute redundancies such that the probability of continuous video playback at all receivers is increased while attempting to maintain a high average per user received video quality.

The proposed algorithm is described below. Consider a scalable video multicast network, similar to the one considered in Section 4.2.2. The received video quality at each user is estimated by substituting the probability of a coded packet reaching each user μ, α_μ instead of α_{avg} in equations (4-4),(4-8),(4-11) and (4-12). Let the estimated video quality at user μ be ϕ_μ. Let the continuous video playback be defined such that the base scalability video layer is received by the receiver node at a probability greater than a predefined probability of θ. The probability of the μth user receiving the base layer, ρ_{1_μ} can be calculated by substituting α_μ instead of α_{avg} in (4-5) and (4-8). Each user is assigned with a score of π_μ such that,

$$\pi_\mu = \begin{cases} \phi_\mu & ; \rho_{1_\mu} > \theta \\ 0 & ; \rho_{1_\mu} \leq \theta \end{cases} \qquad (4\text{-}14)$$

Thus, considering the total number of φ users, the average score obtained by a user, $\bar{\pi}$, is calculated as,

$$\overline{\pi} = \frac{\sum_{\mu=1}^{\varphi} \pi_v}{\varphi} \qquad (4\text{-}15)$$

The optimal $m_1, m_2, \dots, m_z$ values, such that the probability of continuous video playback at all users is increased while attempting to maintain a high average per user received video quality is calculated as,

$$Max\left(\overline{\pi}\right)$$

Such that,

$$\sum_{d=1}^{z} \left(m_d\right) = R \qquad (4\text{-}16)$$

4.3 Results and Discussion

The same simulation system explained in Section 3.2.2 is simulated to estimate the performance of proposed algorithms. Furthermore, the PNC coefficients are compressed using the algorithm proposed in Section 3.2.

4.3.1 Increasing the Probability of Decoding an NC Generation

A 1.35MB file is repeatedly multicast to all users in the simulated network by gradually increasing the per link packet loss rate and thereby decreasing the probability of decoding an NC generation at an average user, α_{avg}. The data packets are organized in to multi-generations where each multi-generation consists of 3 generations with $n_1 = n_2 = n_3 = 50$. Figure 4-2 presents the average probability of decoding an NC generation at users obtained using the proposed method at a 10% redundancy rate. Results of the proposed method are compared against classical NC-MGM. Simulation results exhibit that the proposed method is capable of increasing the average probability of decoding an NC generation by up to 25%. Furthermore, it is observed that the performance gain obtained

by the proposed algorithm increases as the probability of a coded packet reaching the user decreases. Classical NC-MGM evenly distributes redundancies across NC generations in a multi-generation without taking the network status in to consideration. In contrast, the proposed optimization algorithm selects the best redundancy distribution to maximize the probability of decoding NC generations at an average user. This in return increases the probability of decoding an NC generation. The redundancy distribution by the proposed algorithm at different values of α_{avg} is illustrated in Figure 4-3. It can be observed that the proposed algorithm allocates more redundancies to preceding generations in the multi-generation at lower α_{avg} values. This increases the probability of at least decoding the preceding generations when packet losses are high. However, when α_{avg} increases, more redundant coded packets are added to succeeding generations. It should be noted that the redundant packets from succeeding NC generations in a multi-generaton can be used to decode any preceding NC generations. Therefore, when α_{avg} is high, more redundancies are added to succeeding generations to increase the probability of decoding all the NC generations in a multi-generation.

Figure 4-4 illustrates the average of per user probability of decoding an NC generation in a multi-generation at different redundancy rates when $\alpha_{avg} = 0.9$. Accordingly, it can be observed that the probability of decoding an NC generation with the proposed algorithm is consistently better compared to classical NC-MGM. Figure 4-5 presents the redundancy distribution among NC generations in the multi-generation at different redundancy rates for the same value of α_{avg}. The redundancy allocation behavior in this case is similar to that when α_{avg} is increased gradually.
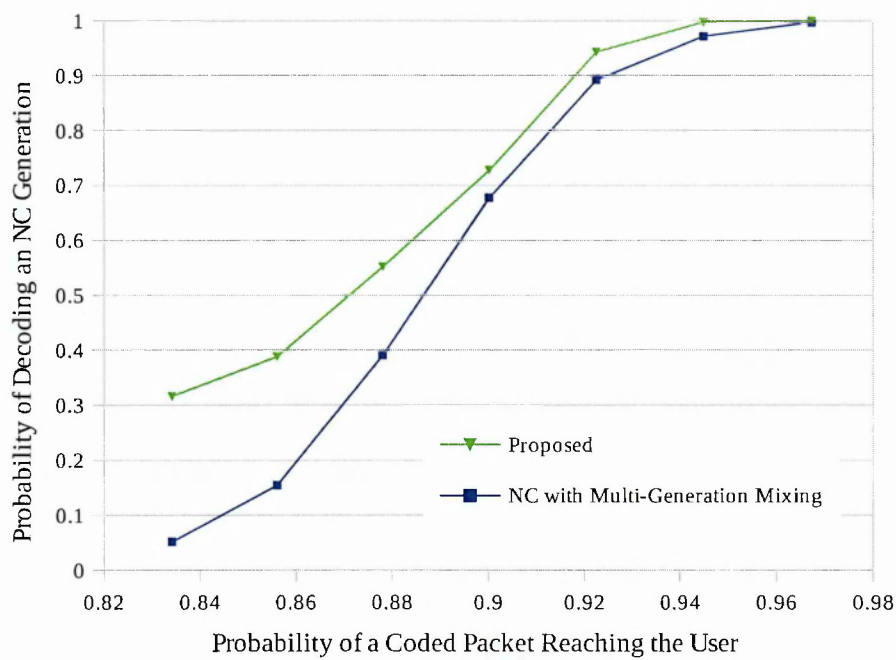
Figure 4-2 Average probability of a user decoding an NC generation against the probability of a coded packet reaching the user.

Figure 4-3 Redundancy distribution among NC generations in the multi-generation against the probability of a coded packet reaching the user.
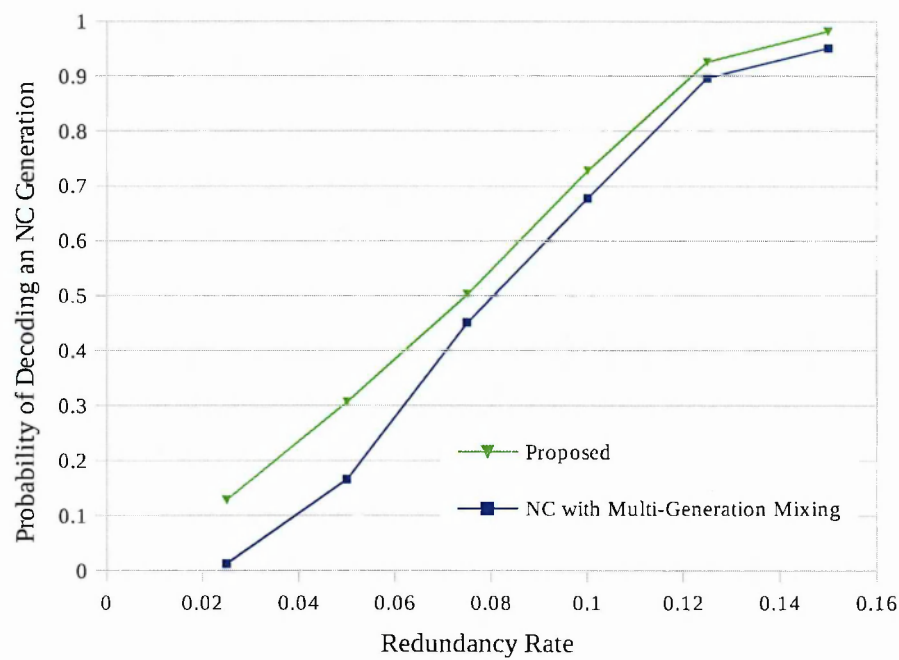
Figure 4-4 Average probability of a user decoding an NC generation against the redundancy rate.
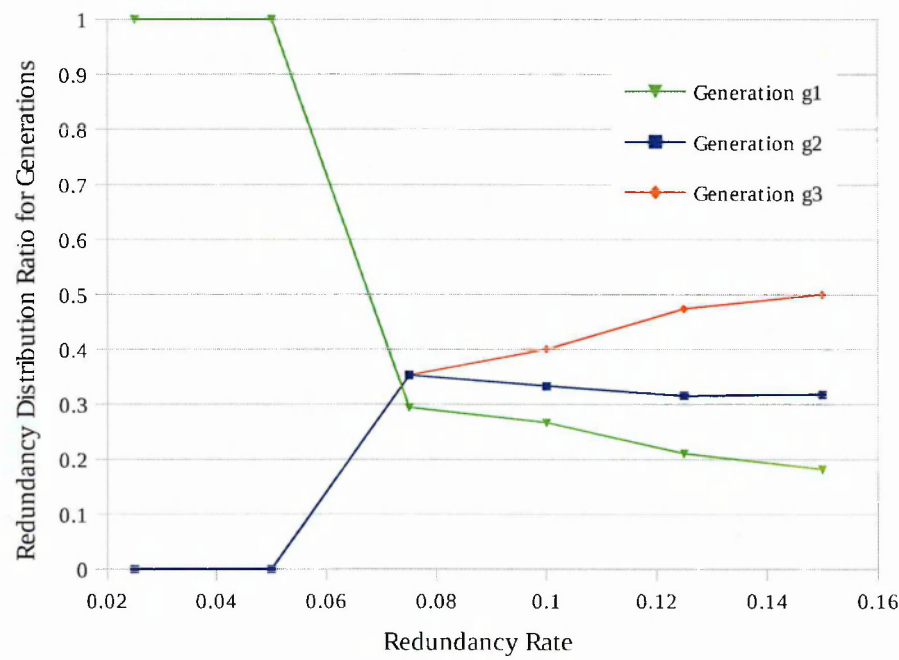
Figure 4-5 Redundancy distribution among NC generations in the multi-generation against the redundancy rate.

Overall, the proposed algorithm is capable of improving the forward error correction capabilities to increase the probability of decoding an NC generation amidst packet losses in a multicast stream. Thus it is capable of utilizing available bandwidth resources efficiently to mutlicast real-time data, such as a video stream, across a network exhibiting packet losses.

4.3.2 Increasing the Received Video Quality

The performance of the algorithm proposed in Section 4.2.2 is assessed by multicasting over 3000 video frames of several scalable video sequences per each simulation, across the simulated network. The redundancy rate for network coding is set to 10%. Resolution of the considered video sequences is 1920x1080. These video sequences are encoded using the JSVM encoder [104] complying to the scalable extension of the H.264/AVC [105] standard with three scalability layers (it should be noted that the proposed algorithms can be used with any video format, which supports scalability layers). These scalability layers are generated by encoding the base layer, enhancement layer 1 and enhancement layer 2 at QP values of 35,30 and 25 respectively. The considered slice size is 1250 bytes. Results are presented for two video sequences, namely, BasketBall Drive and Cactus sequences.

Figure 4-6 illustrate the average of the per user received video quality obtained with the proposed algorithm when BasketBall Drive and Cactus sequences are multicast across the simulated network. Results of the proposed algorithm are compared against two state-of-the-art reference schemes, namely video multicast with NC-MGM [98] and NC with video interleaving [71]. The proposed algorithm exhibits up to 16dB PSNR gains over NC-MGM and up to 5dB PSNR gains over NC with video interleaving. Reasons for the above observations are described below. This gain is achieved by intelligently allocating redundancies between generations in a multi-generation, as illustrated in Figure 4-7, which illustrates the redundancy distribution among NC generations the proposed algorithm for the multicast of BasketBall Drive and Cactus sequences. When the value of α_{avg} is lower, the proposed algorithm increases the amount of redundancies added to generation g_1. Thus, users can at least recover the base layer of a scalable video when the network packet losses are high. Whereas, when α_{avg} increases, more redundancies are added to generations g_2 and g_3 in the multi-generation. Hence, the probability of

recovering more scalability layers in addition to the base layer of the video is increased, which in return increases the received video quality.

It should also be noted that at higher values of α_{avg}, the gain obtained by the proposed optimization algorithm is marginal. This is due to users already receiving video packets at a high probability leaving less room to obtain a significant gain through optimizations. Thus, the proposed algorithm is more useful to multicast scalable video at challenging network conditions where the packet loss rate is high and the available bandwidth is limited. However, the proposed algorithm can also be used to multicast scalable video at a given video quality by using less bandwidth resources compared to the reference schemes by artificially introducing packet drops at the source to reduce the number of packets transmitted.
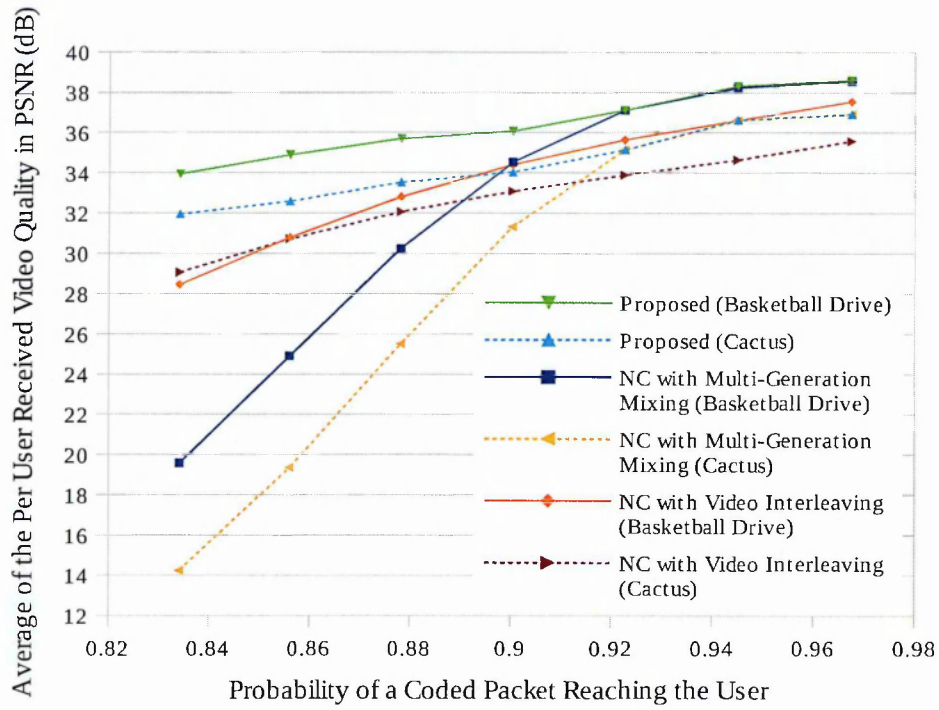
Figure 4-6 Average of the per user received video quality against the probability of a coded packet reaching the average user.
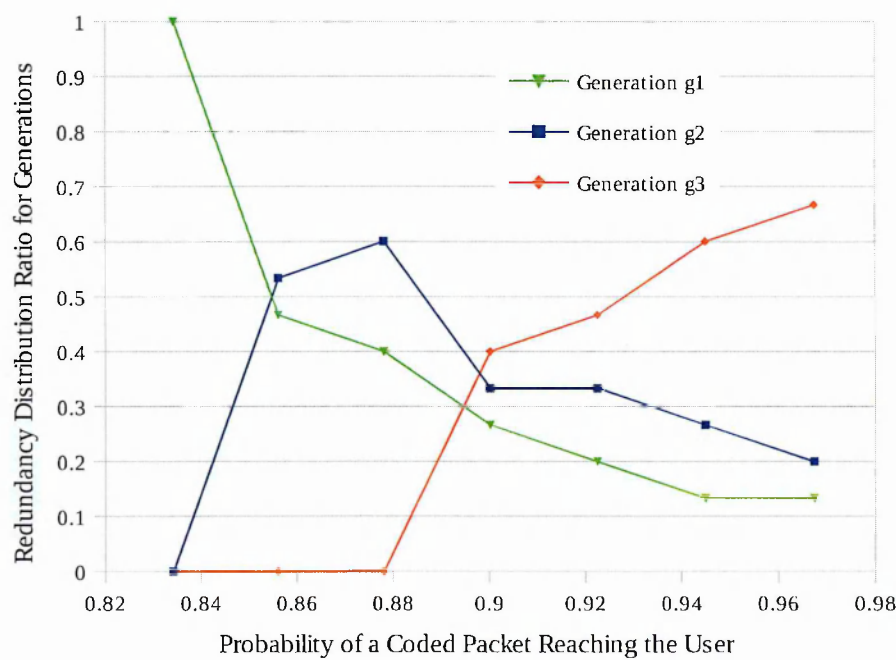
Figure 4-7 Redundancy distribution among NC generations in the multi-generation against the probability of a coded packet reaching the average user.

By means of considering both the end-to-end probability of a coded packet reaching a user and the priority of scalable video layers, the proposed algorithm is capable of allocating redundancies more efficiently such that the estimated video quality of an average user is maximized. Hence, it can be observed that the proposed algorithm efficiently utilizes available bandwidth resources to multicast scalable video at a high quality, across networks exhibiting packet losses.

4.3.3   Increasing the Probability of Continuous Video Playback at All Users

The performance of the algorithm proposed in Section 4.2.3 to increase the probability of continuous video playback at all users is evaluated by multicasting the same scalable video sequences at the same network conditions as described in Section 4.3.2. For this experiment, the required probability of receiving the video base layer, is preset to be $\theta = 0.99$. Figure 4-8 present the probability of receiving the base layer of the video by users when the proposed algorithm is used for multicasting BasketBall Drive and Cactus sequences. Results are compared with the similar probability obtained with the chosen reference algorithms [71] and [98]. Accordingly, it can be observed that the proposed algorithm is capable of increasing the probability of receiving the base layer at all users up to 18% compared to reference algorithms. This gain is obtained by performing per user optimization to allocate redundancies such that the probability of receiving the base layer

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of the video is maintained above the predefined probability θ. Figure 4-9 illustrates the average of the per user received video quality obtained with the proposed algorithm for BasketBall Drive and Cactus sequences. It is observed that the proposed algorithm is capable of not only increasing the probability of continuous video playback at all users but also increasing the average of the per user received video quality in majority of the conditions. However, the average of the per user received video quality obtained with this algorithm is marginally lower compared to the algorithm proposed in in Section 4.2.2 to increase the received video quality. This quality drop occurs due to applying more redundancies to preserve the base layer of the video in order to maintain the predefined probability of continuous video playback at all users. The redundancy distribution among NC generations by the proposed algorithm for the multicast of BasketBall Drive and Cactus sequences are illustrated in Figure 4-10.
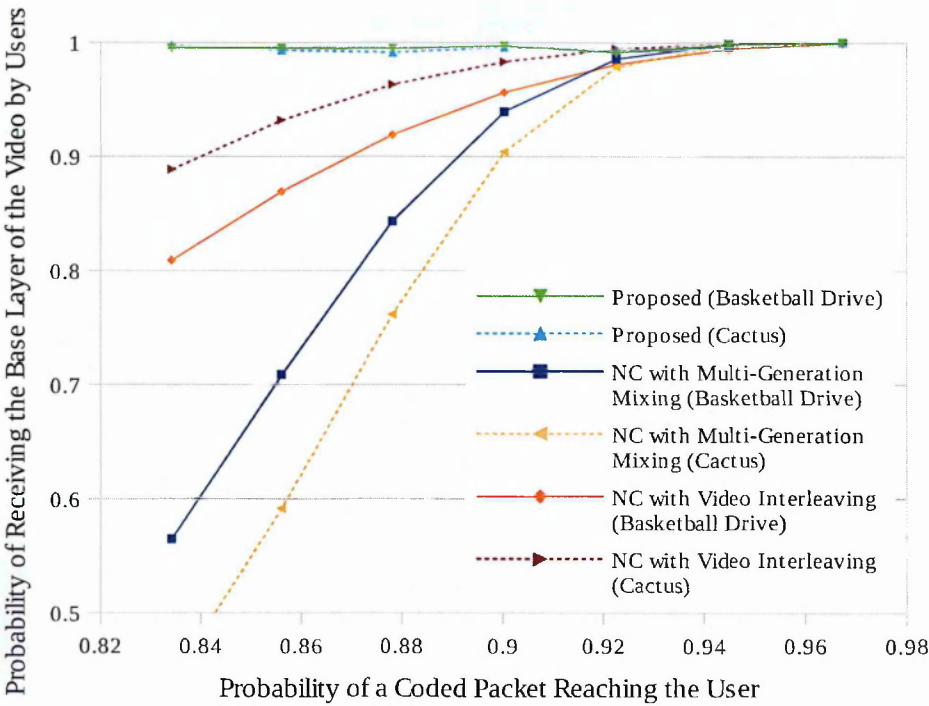
Figure 4-8 The probability of receiving the base video layer vs. the probability of a coded packet reaching the average user.
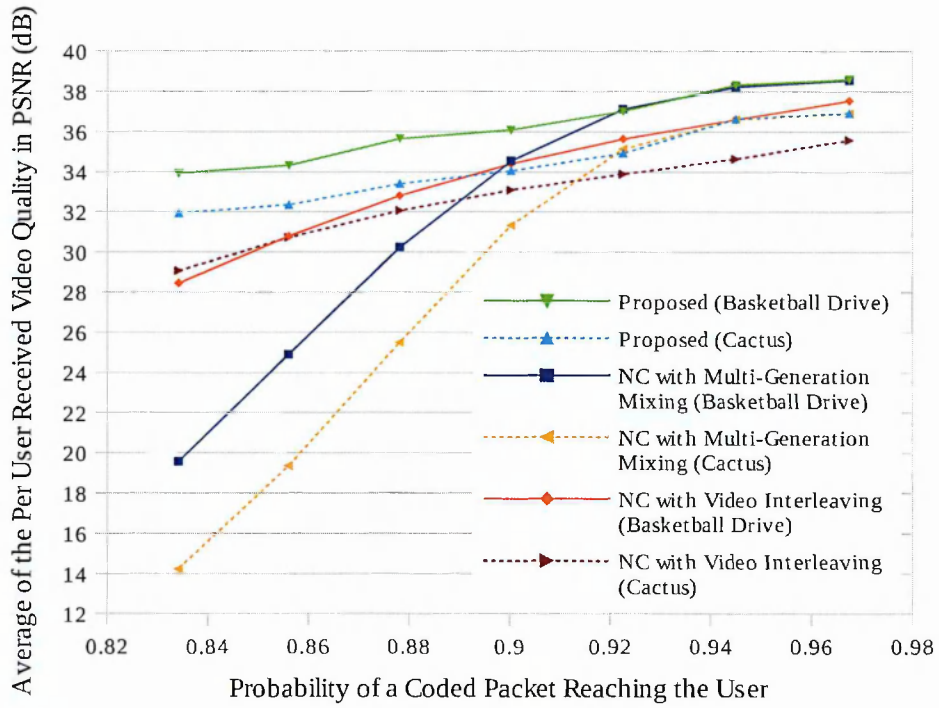
Figure 4-9 Average of the per user received video quality against the probability of a coded packet reaching the average user.
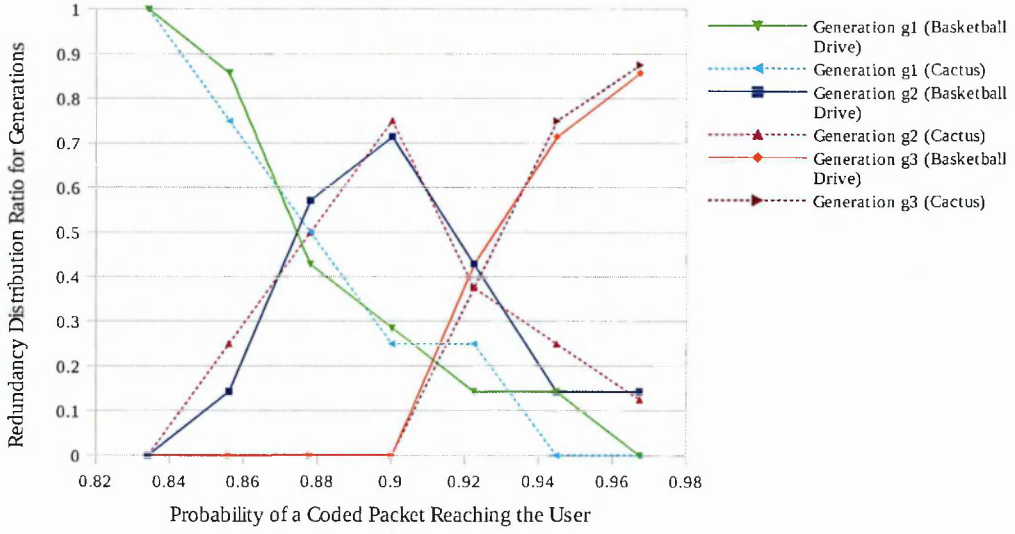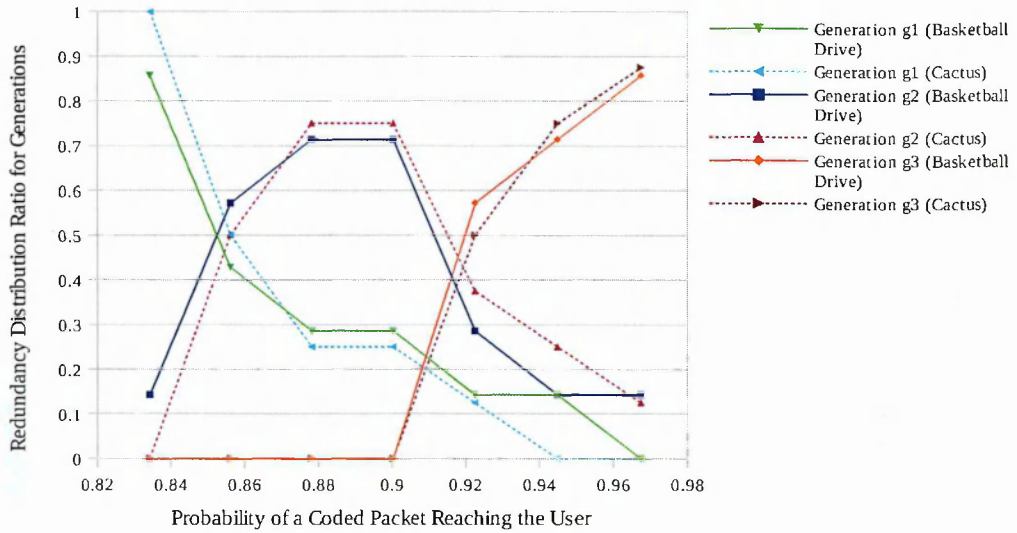

Figure 4-10 Redundancy distribution among NC generations in the multi-generation against the probability of a coded packet reaching the average user.

Hence, it is observed that the proposed algorithm is capable of increasing the probability of continuous video playback of a scalable video mutlicast at all users without

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compromising the average of per user received video quality. Therefore, it is evident that the proposed algorithm is capable of enhancing the user satisfaction of a scalable video multicast across a network that exhibits packet losses.

4.4 Summary

This chapter proposes a set of optimization algorithms to allocate redundancies for NC-MGM to efficiently multicast data and video across networks exhibiting packet losses. The performance of the proposed algorithms is evaluated by means of performing extensive simulations. Up to 25% gains in the probability of decoding an NC generation, 5dB PSNR gains of user received video quality and 18% increase in the average probability of continuous video playback at users in the network are observed compared to state-of-the-art schemes. Hence, by considering both the network status and content properties, the proposed algorithms are capable of efficiently utilizing scarce network bandwidth resources to multicast high quality video content at a satisfactory quality across networks. Thus, the proposed algorithms are useful to cater the increasing demand for real-time high volume video multicast services.

Chapter 5

5 Content and Channel-Aware Channel Coding and Network Coding for Efficient Multicast Over Wireless Networks

5.1 Introduction

State-of-the-art NC algorithms to multicast data and video across wireless networks are briefed in Section 2.5 and Section 2.6.3 in this thesis. Accordingly, it is highlighted that NC algorithms for wireless networks should be capable of increasing the robustness of data and video transmission amidst extremely high packet loss rates in wireless networks. Furthermore, it is noted that the existing algorithms are not capable of allocating cross-layer redundancies efficiently, in order to improve the robustness of a data and video multicast. In response, this chapter proposes content and network-aware cross-layer redundancy allocation algorithms to improve the performance of data and video multicast applications across wireless networks. An overview of the proposed algorithm is illustrated in Figure 5-1. The key contributions in this chapter are highlighted below.

1. Design a channel-aware redundancy distribution algorithm, which intelligently distributes redundancies between channel coding and network coding at individual network nodes in a wireless network, to minimize the redundancies in a data transmission.

2. Optimize the proposed algorithms to increase the quality of a video transmission by considering the delay sensitivity and the priority of video packets.

The rest of this chapter is organized as follows. Section 5.2 describes the proposed algorithms. Section 5.3 presents the performance evaluation of the proposed algorithm, comparing against reference schemes. Section 5.4 summarizes the chapter.
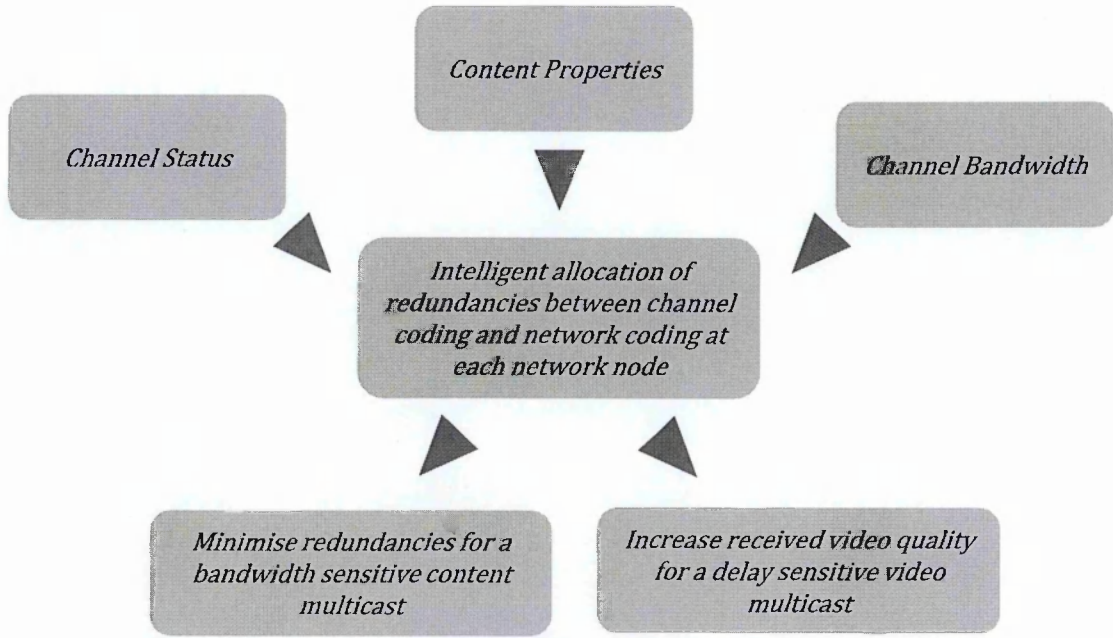
Figure 5-1 Overview of the proposed algorithm.

5.2 System Model

The proposed algorithm is developed considering bandwidth sensitive content and delay sensitive video multicast applications. In case of bandwidth sensitive content multicast applications, the proposed algorithm attempts to minimize the amount of redundancies applied without compromising the robustness of a transmission. As for delay sensitive video multicast applications, the proposed algorithm identifies the content properties of video packets and attempts to use available network resources efficiently in order to ensure the continuous playback of video while increasing the received video quality.

5.2.1 Bandwidth Sensitive Data Multicast Applications

Consider a mesh network consisting of wireless links that exhibits flat-fading where the wireless receptions at different network nodes is independent. The source node builds a shortest-ETX multicast tree [106] based on loss rate measurements, similar to [34] in order to connect the source to all receivers in a network. Next, the source node selects n packets, where each packet is a set of s symbols from a Galois Field of size 2^F. These n packets belong to the network coding generation g. Let i be either the source node or an

intermediate network node and $j_1, j_2, \ldots, j_v$ be v child nodes of node i, in the considered network. π_i denotes the coefficient rank of the received coded packets by node i, from its ancestor nodes. Node i should make sure that all child nodes receive the same amount of information as it has. Hence, in case of no packet errors, the number of coded packets that node i should transmit by performing PNC, y_i, is,

$$y_i = \pi_i - min(\pi_{j_1}, \pi_{j_2}, \ldots, \pi_{j_v}) \tag{5-1}$$

However, since wireless links exhibit packet errors, redundancies should be added to ensure the delivery of at least y_i number of coded packets to all v child nodes. To achieve this with the minimal amount of redundancies, it is proposed to ensure the delivery of at least z_i number of coded packets by all v child nodes at the probability of r_p. The value for r_p is determined empirically such that the total number of redundant bits transmitted is minimized.

Let the average signal to noise ratio at the v child nodes $j_1, j_2, \ldots j_v$ be $\gamma_{j_1}, \gamma_{j_2}, \ldots \gamma_{j_v}$ respectively. Recall node i should make sure that all child nodes receive the same amount of information as it has. Hence redundancies are allocated for FEC considering the child node with the minimum signal to noise ratio, $\check{\gamma}$.

$$\check{\gamma} = min(\gamma_{j_1}, \gamma_{j_2}, \ldots \gamma_{j_v}) \tag{5-2}$$

Let the code rate of channel coding in the physical layer be C_{ch}. Similar to the assumptions followed in [107], for a capacity achieving channel coding scheme (such as Turbo and Low Density Parity Check Coding (LDPC)) the probability of correctly transmitting a packet to the child node with the minimum signal to noise ratio across a Nakagami fading channel, $\check{q}$ is,

$$\breve{q} = \sum_{u=0}^{\kappa-1} \frac{1}{u!} (\kappa\alpha)^u e^{-\kappa\alpha} \tag{5-3}$$

where,

$$\alpha = \frac{(2^{C_{ch}} - 1)}{\gamma} \tag{5-4}$$

and κ is the shape parameter of the Nakagami distribution. It is also worthwhile mentioning that when $\kappa = 1$, the Nakagami-m channel reduces to a Rayleigh fading channel. Subsequently, the probability of correctly transmitting at least y_i packets, r is calculated using the Binomial distribution as,

$$r = \sum_{l=y_i}^{z_i} \binom{z_i}{l} \breve{q}^i (1 - \breve{q})^{z_i - l} \tag{5-5}$$

where, $z_i (\geq y_i)$ is the total number of NC packets (including redundant coded packets, if necessary) generated at node i. It should be noted that PNC requires the transmission of n coefficients per each coded packet. Hence the ratio of $\frac{information\ bits}{total\ number\ of\ bits}$ in an NC packet combination is $\frac{s}{(n+s)}$. Then, the network code rate, C_{nw} can be calculate as.

$$C_{nw} = \frac{y_i}{z_i} \cdot \frac{s}{(n+s)} \tag{5-6}$$

Thus, the total code rate of node i, (C_{t_i}) is,

$$C_{t_i} = C_{ch_i} \cdot C_{nw_i} \tag{5-7}$$

Generally, channel coding schemes are capable of using several pre-defined code rates. Let the optimal values for C_{ch_i} and z_i such that C_{t_i} is maximum be $\acute{C}_{ch_i}$ and $\acute{z}_i$ respectively. $\acute{C}_{ch_i}$ and $\acute{z}_i$ are calculated by,

$$Max(C_{t_i})$$

Such that,   (5-8)

$$r \geq r_p$$

The above optimization is performed using a brute-force analysis. It should be noted that these values depict the optimal distribution of redundancies between channel coding and network coding for the prevailing link condition. The source node keeps on transmitting until the receivers acknowledge the source of sufficient reception of coded packets to decode an NC generation. Upon receiving the acknowledgement from all receivers, node i begins to multicast the consecutive NC generation and flushes packets belonging to the previous generation from its buffer.

5.2.2 Delay Sensitive Scalable Video Multicast Applications

The proposed algorithm for a delay sensitive video multicast is overviewed in Figure 5-2. Accordingly, each network node considers the priority of incoming video packets and network parameters such as channel status and channel bandwidth of child nodes. Then, redundancies are allocated for channel coding and network coding, increasing the probability of delivering at least the high priority scalable video layers.

Similar to the case of bandwidth sensitive content multicast, the proposed algorithm for delay sensitive video multicast initiates by building a shortest-ETX multicast tree to connect the source node to all of the receivers. Subsequently, video packets are linearly coded at the source node, as explained below.

Consider the transmission of a GOP belonging to a scalable video stream containing the base layer and $k-1$ enhancement layers. Let each layer have $n_1, n_2, ... n_k$ number of packets with each packet having s symbols from a Galois Field of size 2^F. Packets

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belonging to the base layer and individual enhancement layers are allocated to k distinct network coded packet generations, $\eta, (\eta + 1), ..., (\eta + k - 1)$. In addition, each coded packet belonging to k scalable video layers are flagged with the priority of the scalable video layer using the content priority class, as illustrated in Figure 5-3. This will allow network nodes to identify the scalable video layer of the video data contained in an incoming NC packet. Accordingly, NC packets combinations containing video packets belonging to the base layer of a video has a content priority class of zero whereas NC packets belonging to the hth enhancement layer has a content priority class of h. After transmitting coded packets belonging to all k generations, the source node begins transmitting coded packets from the consecutive GOP. It should be noted that there is no waiting for receiver acknowledgements in case of a video multicast unlike in a bandwidth sensitive content multicast in order to minimize transmission delay and jitter.

Hereafter, operations performed at the source node and relaying nodes are similar, as explained below. Consider node i, which may either be the source node or a relay node in the considered WMN. Similar to the case of a bandwidth sensitive content transmission, node i is required to transmit y_{i_g} number of coded packets from each generation g, which contains video packets belonging to a particular scalability layer. y_{i_g} is calculated using (5-1). Considering the channel status, node i calculates the optimum number of coded packets $\acute{z}_{i_g}$ and the optimal channel code rate, $\acute{C}_{ch_{i},g}$ for the generation g using (5-8) in such a way that the total code rate, C_{t_i} is maximised while $r \geq r_p$. C_{t_i} is calculated using using (5-8). Similar to the case of a bandwidth sensitive content multicast, a suitable value for r_p is determined empirically such that the received video quality is maximized for a video multicast, as elaborated in Section 5.3.3.2.
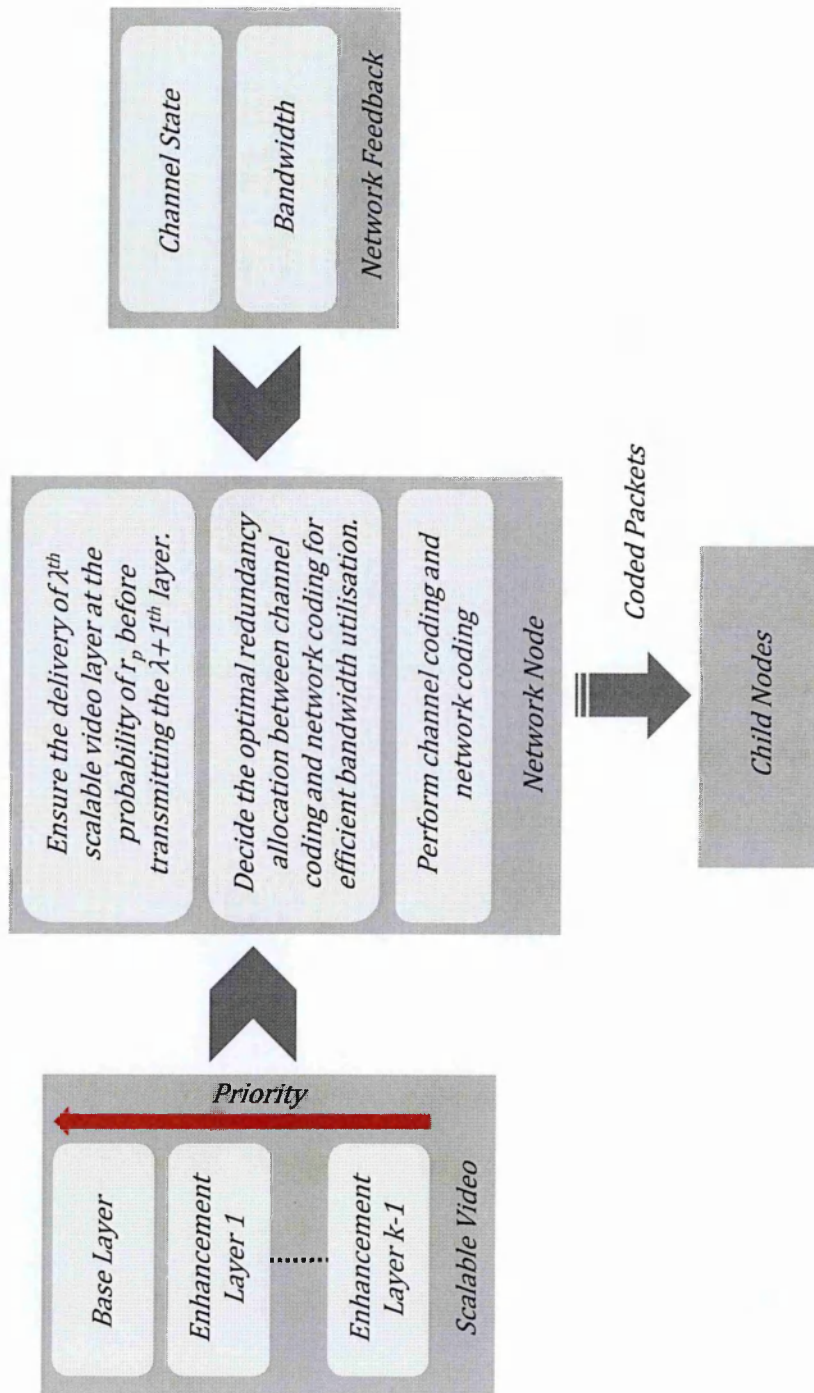
Figure 5-2 Overview of the proposed algorithm for video transmission.

Coded Packets containing Video Base Layer	Content Priority Class 0	Generation η
Coded Packets containing Enhancement Layer 1	Content Priority Class 1	Generation $\eta+1$
⋮		
Coded Packets containing Enhancement Layer $(k-1)$	Content Priority Class $(k-1)$	Generation $\eta + k - 1$

Figure 5-3 Assignment of content priority classes for coded packets from scalable video layers.

However, it should be noted that a video multicast is delay sensitive. Hence, in case the bandwidth available is limited, network nodes should intelligently select the video packets to transmit such that the received video quality of a user is least affected. Let the bandwidth required to transmit y_{i_g} number of coded packets belonging to a generation g be denoted as b_{i_g}. If the channel bandwidth is d_i, the requirement to transmit all scalable video layers with the optimal number of coded packets is,

$$\sum_{g=\eta}^{\eta+k-1} b_{i_g} \leq d_i \tag{5-9}$$

In case (5-9) is satisfied, video packets belonging to all scalable video layers can be transmitted successfully. However, in case (5-9) is not satisfied, node i only transmits coded packet belonging to $w < k$ scalability layers, considering the priority of each scalable video layer. w is calculated such that,

$$\sum_{g=\eta}^{w} b_{i_g} \leq d_i < \sum_{g=\eta}^{w+1} b_{i_g} \tag{5-10}$$

Additionally, the remaining bandwidth (i.e., $d_i - \sum_{g=\eta}^{w} b_{i_g}$) is used to transmit the maximum possible amount of packets from generation $w + 1$. These might be useful in

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case if network nodes and receivers in the downstream receive additional innovative packets by means of opportunistic listening.

5.3 Results and Discussion

5.3.1 Simulation Setup

To evaluate the performance of the proposed method, the proposed method is simulated on a wireless mesh network with 100 static nodes randomly placed in an area of 100 × 100 square units, as illustrated in Figure 5-4. The source node is located at the origin of the graph. The transmission distance of each wireless node is 30 units. Bandwidth of each link is different from others. All wireless links are Rayleigh fading channels. The channel coding scheme considered in this simulation is LDPC with discrete code rates of $\frac{7}{8}, \frac{5}{6}, \frac{3}{4}$ and $\frac{2}{3}$. The Galois Field size for network coding is 2^8, which is sufficiently large enough for practical applications [65], [108]. Each coded packet contains $s = 1400$ symbols from the considered Galois Field.

For bandwidth sensitive content multicast simulations, a scenario in which the source node transmits a 1.35MB file is considered. Similarly, delay sensitive video multicast simulations are performed with high definition video sequences encoded using the scalable extension of H.264/AVC in the JSVM encoder. The video sequences are of 1920 x 1080 pixel resolution. The base layer and two enhancement layers are encoded with QPs 35, 30 and 25, respectively. These values are selected such that each scalable layer has a distinctive difference in bit-rates and quality.

Each frame is divided in to slices of 1400 bytes in order to be accommodated in a network coded packet with 1400 symbols from a Galois Field of size 2^8. Per each simulation, over 3000 video frames are transmitted across the simulated network. PSNR is used to measure the quality of received video sequences.
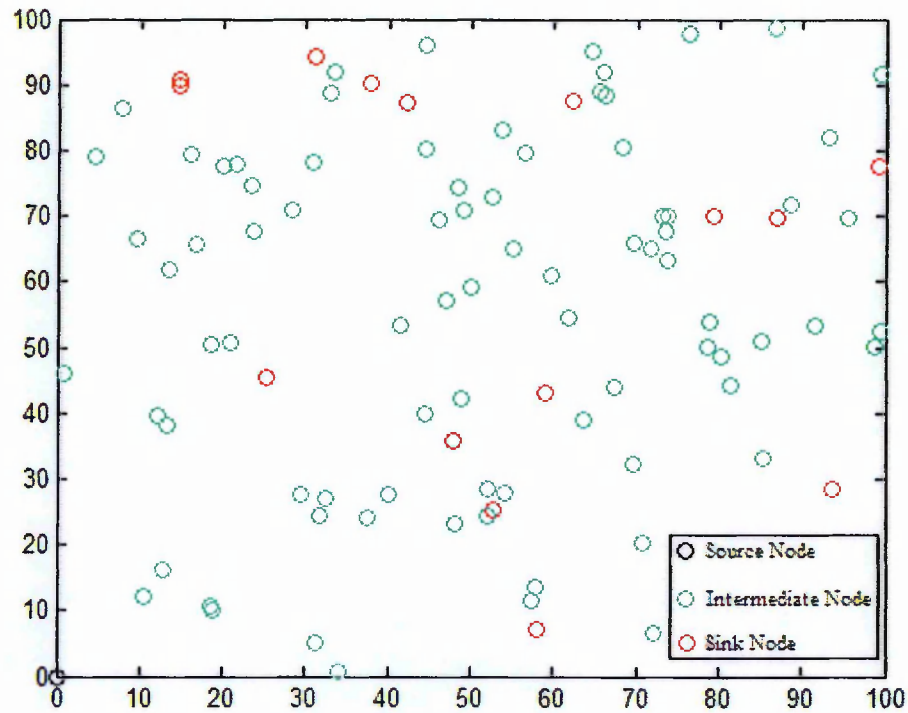
Figure 5-4 Wireless network topology

5.3.2 Evaluation Metrics

Following are the metrics used to evaluate the performance of the proposed algorithms compared to the reference schemes.

5.3.2.1 Bandwidth Sensitive Data Multicast Simulations

The objective of the proposed method for a bandwidth sensitive content multicast is to achieve a 100% packet delivery ratio while minimizing the total amount of redundancies allocated for channel coding and network coding. Under the proposed method each network node allocates redundancies independently. Representing the abovementioned characteristics of the proposed method, following metric is used to evaluate a bandwidth sensitive content multicast.

Average Number of Transmitted bits

$$= \frac{\sum_{\forall network\ nodes} total\ number\ of\ bits\ transmitted}{\begin{matrix} number\ of\ network\ nodes \\ in\ the\ multicast\ tree \end{matrix} \times \begin{matrix} original\ file \\ size\ in\ bits \end{matrix}} \qquad (5\text{-}11)$$

5.3.2.2 Delay Sensitive Video Multicast Simulations

Unlike in a bandwidth sensitive content multicast, the objective of a delay sensitive video multicast is to deliver packets such that the probability of delivering at least the high priority video packets is increased. To achieve this, the proposed method independently allocates channel coding and network coding redundancies at each node such that the receiver video quality is maximized. In order to represent these characteristics, the following metric is used to evaluate a delay sensitive video multicast.

Average Received Video Quality

$$= \frac{\sum_{\forall\ receivers} \sum_{\forall\ video\ frames} PSNR\ of\ received\ video\ frame}{number\ of\ receivers \times number\ of\ transmitted\ video frames} \qquad (5\text{-}12)$$

5.3.3 Simulation Results

5.3.3.1 Bandwidth Sensitive Data Multicast Simulations

The objective of a bandwidth sensitive content multicast is to achieve a 100% packet delivery ratio by means of either FEC or ARQ by transmitting minimal number of redundant bits.

Initial simulations are performed to determine an empirical value for r_p. Figure 5-5 illustrates the average number of bits transmitted with the proposed algorithm vs. r_p for average signal to noise ratio values of 10 and 20. This shows that the average number of transmitted bits gradually decreases with r_p until a minimum and increases thereafter. The above observations can be explained by considering (5-5) and (5-6). Accordingly, smaller the value of r_p is, smaller the value of C_{nw} becomes. Hence less redundancies are added per transmission. On the other hand, a lower r_p results in a lower probability of accurately

receiving coded packets from node by its child nodes. This in return reduces the end-to-end probability of successfully transmitting a coded packet from the source node to all receivers. Since the source keeps on transmitting coded packets until all the receivers acknowledge, a lower probability of successful transmission of a coded packet increases the number of retransmission attempts. Compromising the amount of redundancies added per transmission and the number of retransmission attempts, an empirical value for optimum r_p can be determined in such a way that the total number of bits transmitted is minimised. Based on the experimental results presented in Figure 5-5, $r_p = 0.992$ is chosen as the optimum value. This value is used for subsequent experiments.

The subsequent set of experiments compare the performance of the proposed method against the reference scheme [34]. Since [34] does not specify a channel code rate, the reference technology is simulated for all considred channel code rates. Figure 5-6 illustrates the average number of bits transmitted vs. the average signal to noise ratio. According to Figure 5-6, it can be observed that the proposed algorithm can significantly reduce the average number of transmitted bits compared to the reference scheme. Unlike the proposed algorithm, [34] does not estimate the redundacnies that should be allocated for channel coding and network coding to minimise the average number of transmitted bits. As a result, the performance of the proposed algorithm improves over the reference scheme.

Furthermore, the end-to-end probability of successful transmission of a coded packet increases rapidly with the number of hops when the per-hop packet delivery probability is lower. However, since the proposed method ensures a high per-hop packet delivery probability, it is more scalable for the size of the network compared to the algorithm proposed in [34]. This argument is supported by the results presented in Figure 5-7, which illustrates the average number of bits transmitted vs. the number hops.

5.3.3.2 Delay Sensitive Scalable Video Multicast Simulations

An extensive simulation study is performed to evaluate the performance of the proposed algorithm using a number of video sequences. For the demonstration purposes, results are presented for Basketball Drive and Cactus sequences (refer to Section 4.3 for more information about these sequences). Similar to the case of bandwidth sensitive content multicast, a suitable r_p value for delay sensitive video multicast applications is determined empirically such that the average received video quality is maximised.
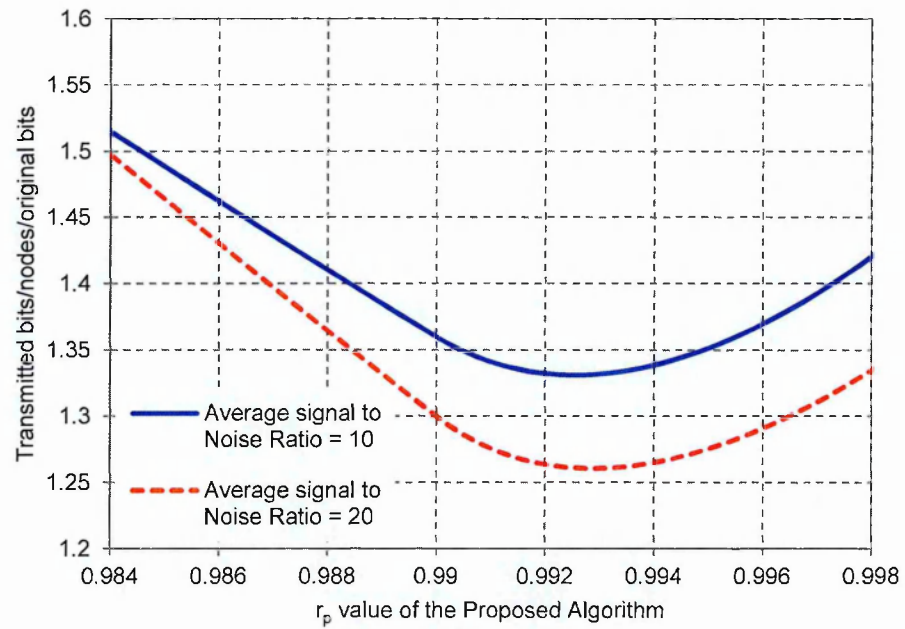
Figure 5-5 Transmitted bits/nodes/original bits vs. r_p value of the proposed algorithm
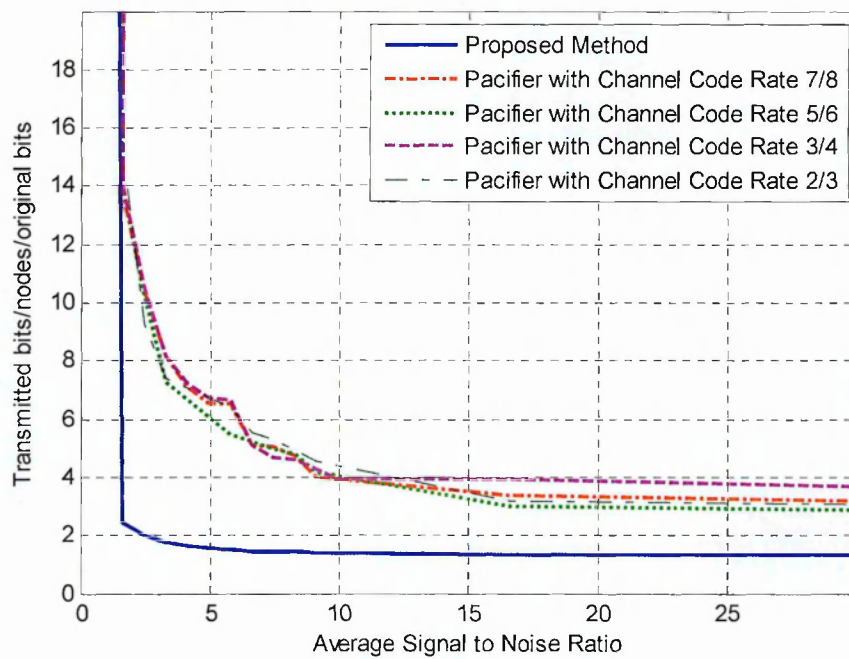
Figure 5-6 Transmitted bits/nodes/original bits vs. average signal to noise ratio.
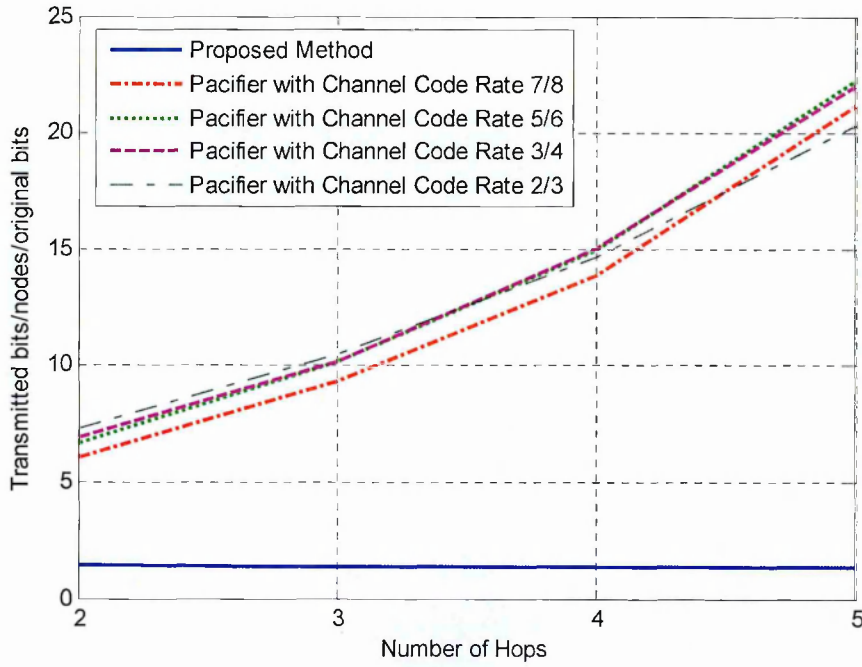
Figure 5-7 Transmitted bits/nodes/original bits vs. number of hops.

Figure 5-8 and Figure 5-9 illustrate the average received video quality of Basketball Drive and Cactus sequences for the proposed method vs. r_p for average signal to noise ratio values of 10 and 20. The received video quality for the sequences over the simulated range of signal to noise ratios is maximised when $r_p = 0.9999$. Hence this value is selected for further simulations presented in this section.

It should be noted that the optimum r_p value for delay sensitive video multicast applications is greater than that for bandwidth sensitive content multicast applications. The only error correction mechanism applied for delay sensitive video multicast is FEC due to strict delay constraints. Hence, redundacnies are applied for delay sensitive video multicast applications to ensure a higher r_p value compared to bandwidth sensitive content multicast applications.
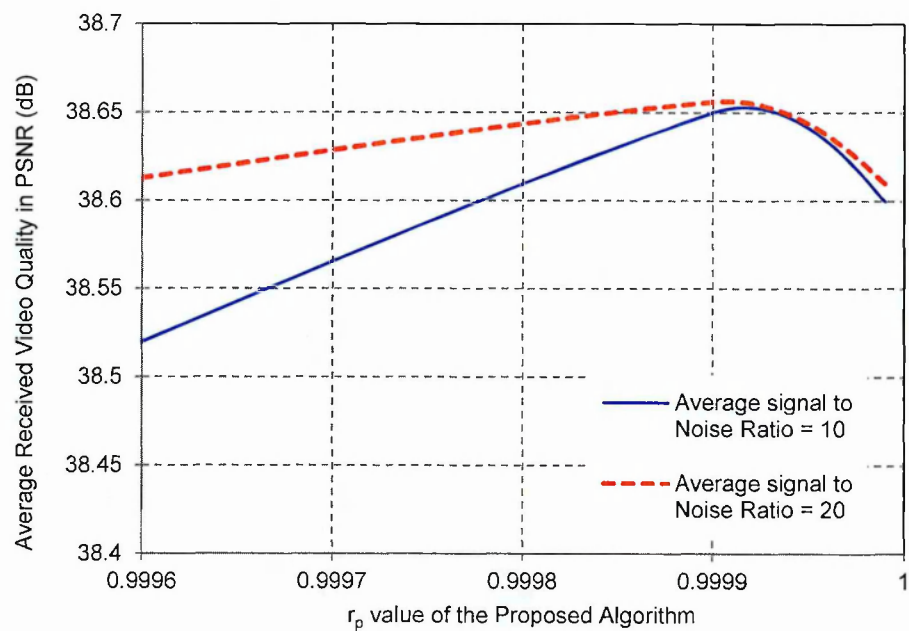
Figure 5-8 Average received video quality vs. r_p value of the proposed algorithm for the Basketball Drive sequence.
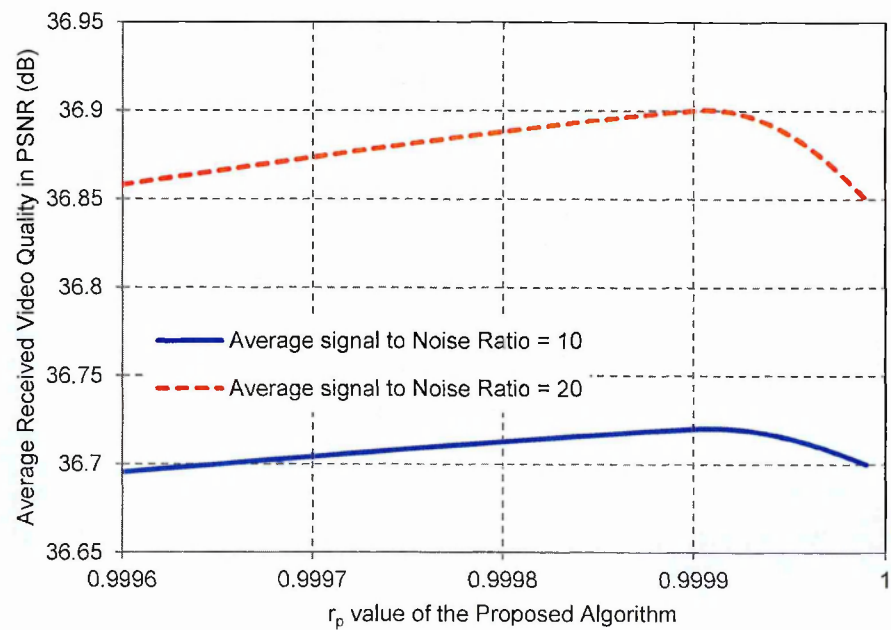
Figure 5-9 Average received video quality vs. r_p value of the proposed algorithm for the Cactus sequence.

The next set of experiments are performed in order to compare the performance of the proposed algorithm against the reference scheme Distributed Robust Optimization for Scalable Video Multirate Multicast (DROVMM) [99]. Due to similar reasons as discussed in Section 5.3.3.1, the algorithm presented in [99] is simulated for all the considered channel code rates. The maximum backup percentage parameter, B_d, in the reference technique is to $B_d = 0.5$ to offer a higher protection level [99]. Figure 5-10 and Figure 5-11 illustrate the average received video quality vs. average signal to noise ratio for Basketball Drive and Cactus sequences respectively. The obtained simulation results show that the proposed technology consistently outperforms the reference technique. The proposed algorithm allows individual network nodes to independently allocate redundancies considering the content priority and link condition. In contrast, the maximum backup percentage, B_d in [99] is fixed at the source. Hence, the network code rate remains constant throughout the network regardless of the differences in network codition at different links. Furthermore, the proposed method allocates redundancies more
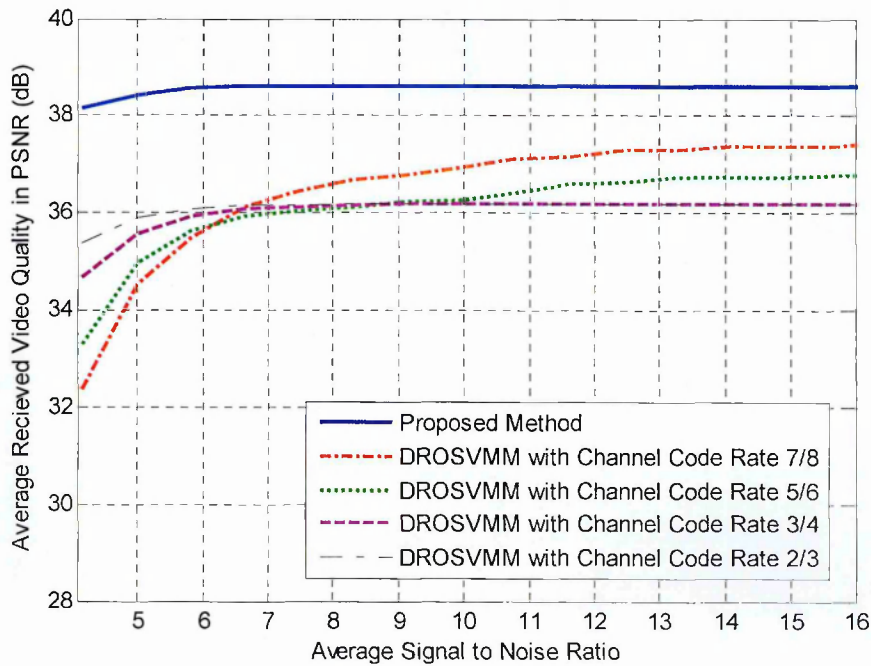
Figure 5-10 Average received video quality vs. average signal to noise ratio for the Basketball Drive sequence.
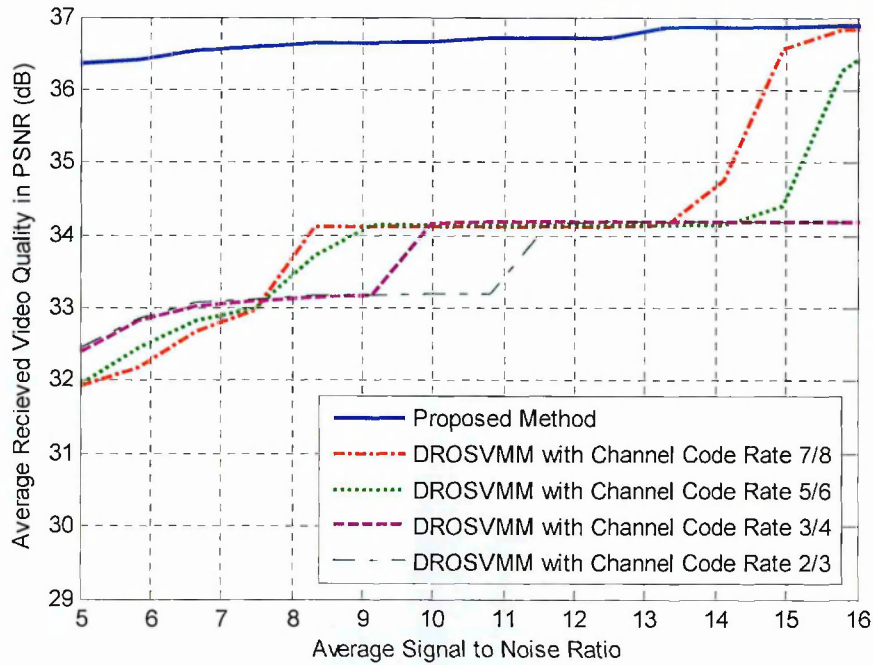
Figure 5-11 Average received video quality vs. average signal to noise ratio for the Cactus sequence.

intelligently for channel coding and network coding in such a way that it suits the best for each network link for a scalable video transmission, unlike the reference technique.

5.4 Summary

This chapter proposes a novel content and network-aware algorithm for network nodes to independently allocate cross-layer redundancies in order to increase the robustness of a data and video multicast across a wireless network. The objectives are to minimize the redundancies for bandwidth sensitive multicast applications and increase the received video quality for delay sensitive video multicast applications. Simulation results for bandwidth sensitive content multicast applications indicate up to $10 \times$ reduction in the number of redundant bits transmitted. Up to 3.5 dB gains in user received video quality is observed for delay sensitive video multicast applications. Hence the proposed method can be used to efficiently utilize wireless resources for multicast applications in packet loss prone WMNs.

What is more, the proposed algorithm for delay sensitive multicast applications is not only limited for scalable video transmission but can be explored for other forms of video transmission where video content can be classified in to multiple content priority classes.

Chapter 6

6 Joint Source and Network-Coding for Error Resilient Video Multicast Applications

6.1 Introduction

Chapter 5 of this thesis proposes a joint optimization of channel coding and network coding to improve the performance of data and video multicast applications. However, the opportunities of exploiting video coding techniques to improve the video quality of video multicast applications in PNC enabled networks are not explored. In response, this chapter analyses the effect of using redundancies for source video coding and PNC, on the error resilience performance of high volume video multicast applications. The objective is to explore the opportunities of performing joint source and network coding to improve the video quality of high volume video multicast applications across networks with packet losses.

One of the major challenges in real-time high volume video multicast is to combat network packet losses, as retransmission using ARQ is not feasible due to strict delay constraints. The conventional method for reliable video multicast over packet lossy networks is to transmit redundant video data along with main video streams. This increases the probability of receiving sufficient compressed video data to decode the original video at a satisfactory quality. However, as the bandwidth requirements to multicast high volume video content increases, it is essential to investigate bandwidth efficient means of adding redundancies to combat packet losses.

As explained in Section 2.3, PNC is one of the promising technologies that have proven ability to enhance not only the throughput but also robustness and reliability of existing

network infrastructures. Redundancies for PNC are generated as redundant packet combinations at a layer between the transport layer and the network layer in the OSI model. These redundant packet combinations can be efficiently utilized to increase the robustness of video multicast applications in packet lossy networks. PNC uses redundant packet combinations to recover any lost video packet from a set of video packets grouped in to an NC generation. Hence, redundant NC packets can have an impact on more than one data packet. Whereas any given application layer redundant video packet is useful only if its corresponding packet is lost. As a result, it is envisaged that the same level of redundancies added during network coding is more efficient than adding redundant video data in the application layer in preserving the video quality of a high volume video multicast across networks exhibiting packet losses.

However, there is no significant research performed to allocate redundancies for the joint optimization of source (video encoder) and network coding to improve the error resilience of high volume video multicast applications. In response, this chapter performs a simulation study to investigate efficient means of allocating cross-layer redundancies for high volume video multicast applications. The rest of this chapter is organized as follows. Section 6.2 presents the simulation methodology. Section 6.3 presents and discusses the simulation results. Section 6.4 summarizes the chapter.

6.2 Simulation Methodology

This simulation study compares the error resilience performance of state-of-the-art redundant video coding [109] against PNC to multicast high volume video across networks exhibiting packet losses. The objective is to identify efficient means of allocating cross-layer redundancies to increase the robustness against packet losses and thereby improve the quality of high volume video multicast applications.

6.2.1 Network Simulation Setup

A network topology similar to that of WirelessHART [110] is simulated to analyze the effect of adding redundancies either as redundant packet combinations for PNC or as redundant video data in improving the error resilience of high volume video multicast applications. Figure 6-1 and Figure 6-2 illustrate the simulated network topology in multipath multicast and PNC configurations, respectively. In the simulated network, the

source node, node 0, multicasts information to users located at node 7, node 12 and node 14. Accordingly, for the selected network topology, it is observed that all receivers are capable of receiving video packets at the network capacity as defined by the max-flow-min-cut theorem, regardless of the considered routing algorithm.

The capacity of each link is assumed to be 5Mbps. Packet loss patterns for each link is assumed to follow the Gilbert Elliot model with an average packet loss burst length of 2 packets. The considered link packet loss rates are 0%, 0.5%, 2.5% and 4.5%. In case of traditional video multicast without PNC, multicast trees are selected using the nearly disjoint tree construction algorithm [111]. This facilitates the transmission of different streams across nearly disjoint multi-paths in order to increase the loss resiliency. Figure 6-1 illustrates the simulated network in multicast configuration. 3 nearly disjoint trees are constructed (since the min-cut of the network is 3 for all receivers), therefore, different streams can be multicast in different trees. The same network in PNC configuration is illustrated in Figure 6-2. As for network coding simulations, a PNC scheme with a generation size of $n = 50$ is selected. A Galois field size of 2^8 is selected to ensure that the probability of failure to decode network coded packets is negligible [65], [108].

6.2.2 Data Stream Generation for Simulations

The aim of this study is to compare the error resilient performance of PNC with redundant video coding for high volume video multicast applications. In other words, the effect of adding redundancies at application layer is compared against the redundancies applied at the network layer, on the overall performance of error resilient video multicast. This section provides information on how the data streams are generated and simulations are performed to arrive at conclusions. The data stream that is intended to transmit from the sender to the receiver is an encoded stream of stereoscopic video. This study considers 2 High Definition (HD) stereoscopic video sequences, namely GT Fly and Band. To generate the main stream of data, the stereoscopic video is encoded utilizing the version 15.1 of the reference software for the H.264/AVC video coding standard. The left and right views of the stereoscopic video sequence are encoded independently at the same target bit rate. The encoding parameters used for compression of the two sequences are given in Table I.
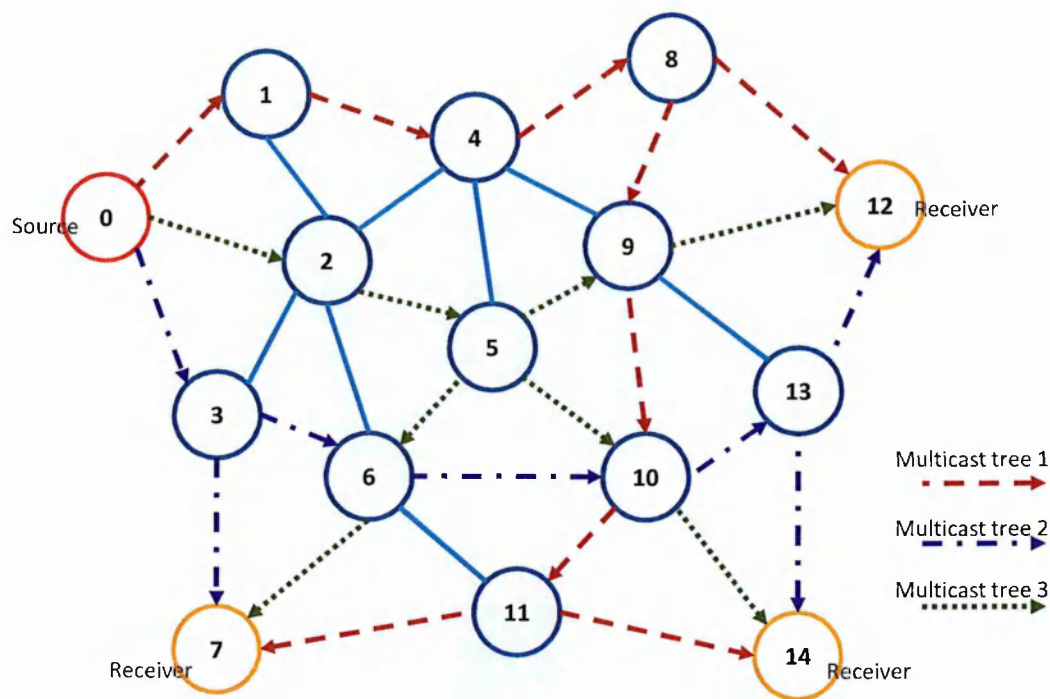
Figure 6-1 Simulated network in multicast configuration
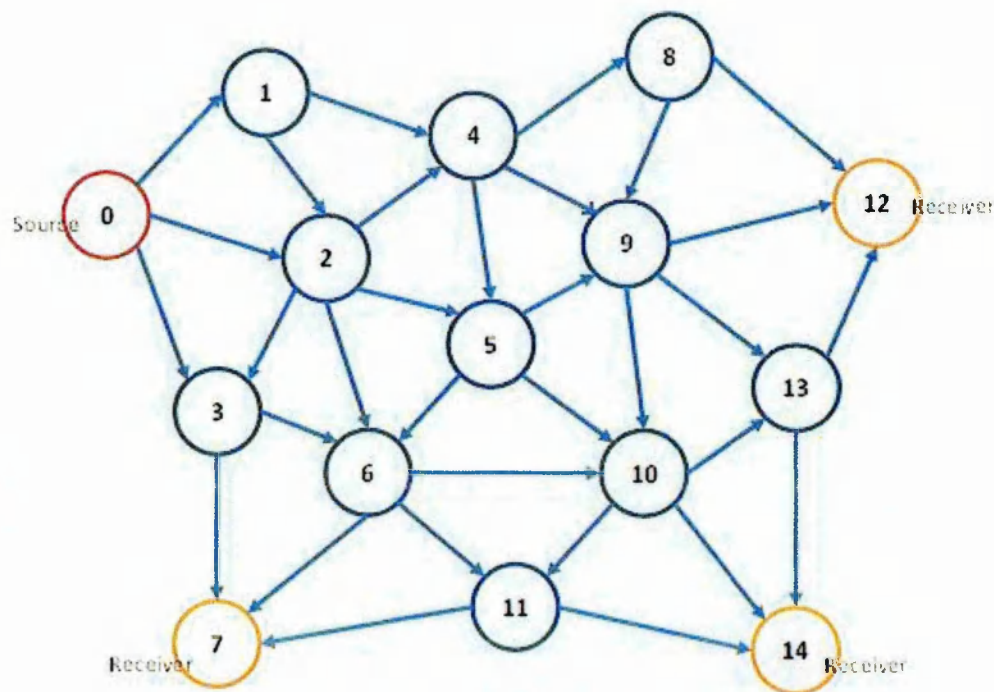
Figure 6-2 Simulated network in the PNC configuration

Table 6-1 Encoding parameters for video compression

Parameter	Value
Number of Frames	125
Profile IDC/ Level IDC	100/40
Intra Period	25
GOP Structure	IPPP…
Entropy Coding	CABAC
Slice Mode/Argument	1400 Bytes/Slice
Rate Control	Enabled, Basic Unit=16
Motion Estimation	Enhanced Predictive Zonal Search

6.2.3 Different Simulation Criteria

6.2.3.1 Configuration A: Multicasting with Redundant Video Streams

In addition to the main stereoscopic video stream, redundant pictures are encoded at specific intervals to be transmitted over different channel conditions. These redundant pictures are then used at the decoder to conceal the video frames that are corrupted due to packet losses. Therefore, the redundant pictures transmitted along with the main content improve the error resilience of the stereoscopic video stream. Thus, the stream of redundant pictures acts as an error resilient tool at the application layer. The amount of redundancy in the application layer can thus be controlled by selecting appropriate intervals at which redundant pictures are encoded. When redundant pictures are encoded at small intervals, the bit rate consumed by the stream of redundant pictures is increased. In the simulations performed in this chapter, the video streams are encoded such that the main left and right views consume a bit rate of 5Mbps each while redundant pictures for the left and right views consume a bit rate of 0.75Mbps each at a 13% redundancy rate and 1.25Mbps each at a 20% redundancy rate.

6.2.3.2 Configuration B: PNC with Redundant Packets at Network Layer

The performance of application layer redundancies (as redundant pictures) is compared against network layer redundancies in terms of error resilience. Redundancies in the network layer are added as redundant packet combinations based on PNC concepts. In this case, the main stereoscopic video is encoded at 10Mbps (5Mbps each for left and right views) without adding redundant pictures. Instead, redundant packets combinations are added at the network layer to account for the considered redundancy level.

6.2.3.3 Configuration C: Network Coding with Redundant Video Stream

Configuration C considers the same video streams described in configuration A (described in Section 6.2.3.1). However, instead of multicasting video packets using the algorithm proposed in [111], PNC is used to multicast video packets. No PNC redundant packet combinations are transmitted. Hence, this configuration has the advantage of using multipaths efficiently similar to that of configuration B. However, there are no redundant NC packet combinations to aid recovering NC packet combinations. Instead, redundant pictures will aid to recover lost video packets.

6.3 Results and Discussion

125 frames of each video sequence (GT Fly Sequence and Band Sequence) are transmitted across the simulated network. Simulations are run 10 times for each considered packet loss rate. Figure 6-3 illustrates average PSNR values of the received GT Fly and Band sequences, when simulation 1 is performed with configurations A, B and C as stated in Table 6-2. In case of configuration A, 13% of the total amount of data transmitted from the source account for redundant video data. In configuration B, PNC is performed with a redundancy rate of 13%. As for configuration C, 13% of the data transmitted from the source are redundant video data. According to Figure 6-3, a considerably higher received video quality is observed with configuration B compared to configuration A for GT Fly and Band sequences up to packet loss rates of 2% and 2.5% respectively. Furthermore, it is observed that the received video quality at 0.5% of packet losses is equivalent to the received video quality when no packet losses were present.

However, for link packet loss rates beyond 2%, the video quality of GT Fly sequence with configuration B becomes inferior to that of configuration A. In case of the Band sequence, the received video quality with configuration B remains similar to the quality obtained with configuration A beyond link packet loss rates of 2.5% up to 4.5%. It should be noted that the quality drop observed with the GT Fly sequence is more significant compared to the Band sequence due to the more complicated texture and movement in the GT Fly sequence.

Considering the results obtained for configuration C, it is observed that when no redundant packet combinations are transmitted, PNC has significantly more packet losses compared to the multipath multicast scenario. This occurs as PNC fails to completely decode a generation even if a single packet combination is lost. Thus, in configuration C, the decoder fails to decode the video sequences when packet losses occur.

Table 6-2 Simulation configurations

Simulation	Configuration	Video Coding Redundancy Rate	PNC Redundancy Rate	Routing Algorithm
1	A	13%	0%	Multi-path Multicast
	B	0%	13%	PNC
	C	13%	0%	PNC
2	A	20%	0%	Multi-path Multicast
	B	0%	20%	PNC
	C	20%	0%	PNC
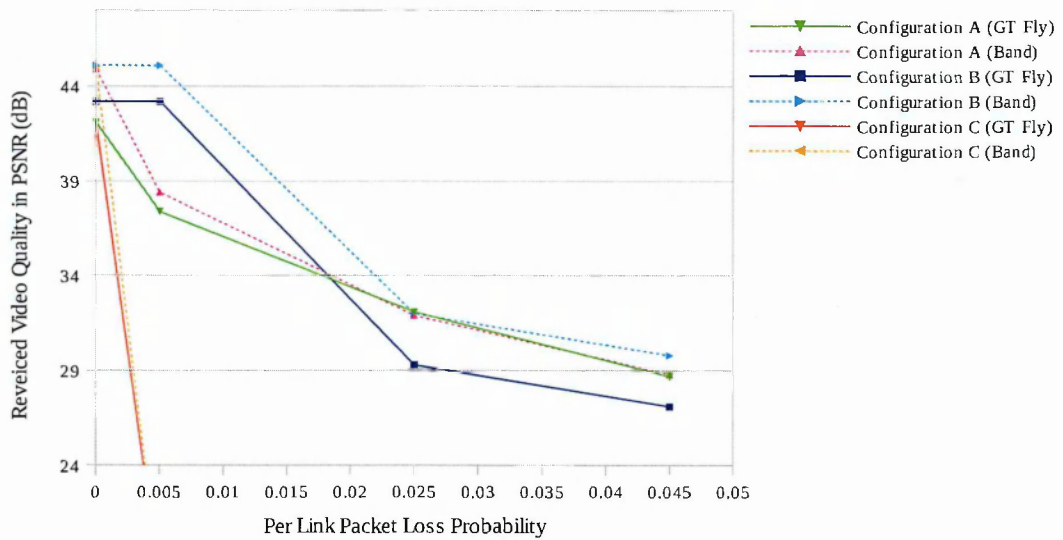
Figure 6-3 Received video quality vs. per link packet loss probability for Simulation 1.

To further investigate on how PNC performs with more redundancies level compared to video multicast with redundant pictures, simulation 2 is performed with the simulation configurations stated in Table 6-2. The results are indicated in Figure 6-4. Accordingly, a significantly better received video quality is observed with configuration B for both GT Fly and Band sequences throughout the simulated range of link packet loss rates, compared to configuration A. Unlike in configuration A, which uses nearly disjoint multicast trees to transmit distinct streams, configuration B is able to use all the network links more efficiently by flooding linear packet combinations in to the forward stream. Since any of these packet combination can contribute to recover any lost packet combination from the same generation at a high probability (since a sufficiently large 2^8 Galois field size is considered in PNC under this context), configuration B excels in utilizing available multi-paths more efficiently than transmitting video data across nearly disjoint multicast trees, in the simulated network. Regardless of the particular combinations lost or the path which introduces packet losses, receivers can fully recover the original video data from a generation once n number of linearly independent packet combinations are received. This characteristic is reflected as the received video quality at 0.5% of packet losses is equivalent to the received video quality when packet losses were not present. In configuration A, when video redundancies are added in the application layer and multicast across nearly disjoint multicast trees, the decoder attempts to conceal

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the lost video information using the redundant video information. However, under this simulation scenario, it is observed that at a redundancy level of 20%, the capability of the decoder to conceal lost video data using redundant pictures is inferior to the capability of network coding to recover the original data using redundant packet combinations. The least resilience against packet losses is observed with configuration C, similar to the observations made with simulation 1.

It can be concluded that when sufficient redundancies can be allocated for PNC to compensate packet losses, using PNC together with PNC redundancies is more suitable in delivering stereoscopic 3D video at a high quality across a packet lossy networks, compared to traditional multicast together with video redundancies. Furthermore, with more redundancies, PNC becomes more capable of concealing packet losses. This qualifies PNC as the better solution for error resilient transmission of 3D video at a high quality across a network exhibiting packet losses. However, it is observed that PNC fails under packet losses when no redundant packet combinations are introduced and the packet losses are severe such that the redundant video data are not capable of successfully concealing the packet losses (such as in configuration C). Thus, using PNC with video redundancies added in the application layer is not a suitable solution.

Furthermore, it is observed that video redundancies applied at the application layer is not capable of recovering complete losses of NC generations occurred when users receive insufficient packet combinations to decode an NC generation. Equally, when sufficient redundancies are available for PNC to recover complete NC generations, applying video redundancies at the application layer is a waste of bandwidth resources. Hence, it is observed that existing video coding techniques are not optimal for video multicast applications in PNC enabled networks. However, the techniques in Chapter 4 and 5 overcome this issue by considering video properties to perform content aware network coding. Yet, it is envisaged that by performing NC aware video coding, it will be possible to jointly optimize source, channel and network coding to efficiently utilize network resources and deliver video at an even higher quality compared to the presented algorithms.
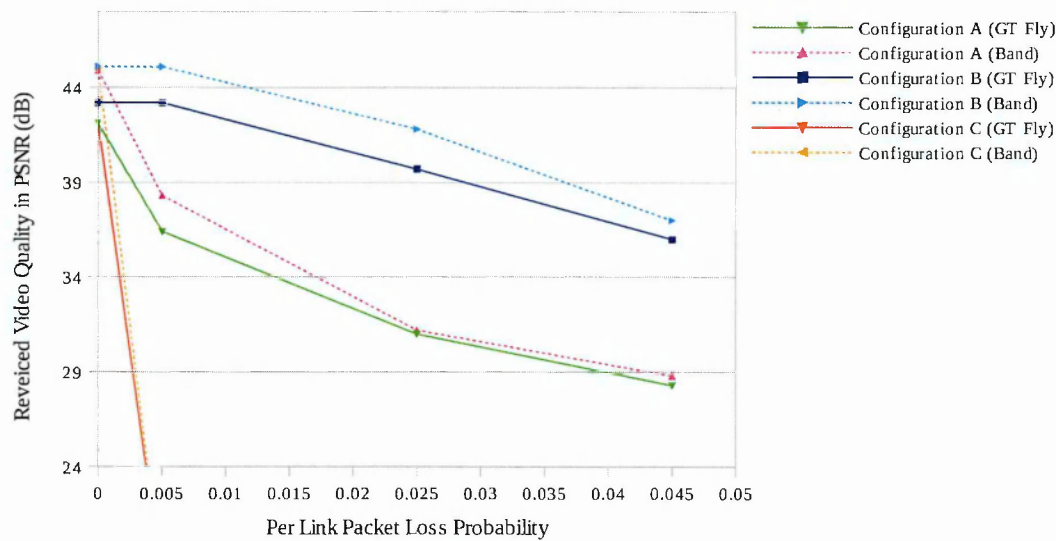
Figure 6-4 Received video quality vs. per link packet loss probability for Simulation 2.

6.4 Summary

Chapter 6 compares the effect of adding video redundancies at the application layer against generating redundant NC packet combinations at the network layer, on the overall performance of error resilient high volume video multicast applications. This is performed by means of an extensive simulation study, which multicast stereoscopic 3D video across a multicast network. Simulation results indicate up to 8dB PSNR gains when redundancies are applied as NC packet combinations compared to applying video redundancies at the application layer along with multipath multicast as the multicast scheme. This qualifies PNC as a better technology to multicast high volume stereoscopic 3D video when video quality is a paramount. However, it is also observed that applying video redundancies with PNC as the multicast scheme is not suitable for video multicast applications. This is due to incapability of video redundancies to conceal the complete loss of NC generations when sufficient PNC redundancies are not available to recover packet losses. Hence, it is observed that existing video coding techniques are not optimal for video multicast applications in PNC enabled networks. These results pave the path for future research directions to design NC aware video coding algorithms. Therefore, the performance of video multicast algorithms proposed in Chapter 4 and Chapter 5 can be

further improved by means of performing joint optimization of source-channel and network coding.

Chapter 7

7 Conclusions and Future Work

7.1 Preamble

This thesis presented various algorithms and experimental investigations to improve the performance of a multicast network using network coding, especially for bandwidth hungry video multicast applications. The achievements of the work presented in this thesis and contributions to the knowledge are presented in Section 7.2. Subsequently, the relevant future work that could be carried out to further develop the conclusions of this thesis is presented in Section 7.3.

7.2 Conclusions

The research work presented in this thesis proposes novel NC algorithms, which efficiently utilize network resources to increase the performance of data and video multicast applications in NC enabled networks. Following is a concluding summary of the research objectives achieved throughout the work presented in this thesis.

The first part of this thesis proses several novel NC algorithms, which increases the network throughput by minimizing the NC coefficient overhead and improving the robustness of a transmission against packet losses. A novel algorithm is proposed to minimize the coefficient vector overhead in NC, which in return improves the network throughput up to 15% compared to state-of-the-art reference schemes. In addition, a method to estimate the probability of decoding an NC generation is proposed to allocate redundancies for NC to achieve either a predefined packet delivery probability to meet a given QoS requirement or to optimize the throughput to maximize network resource utilization. The proposed method is capable of multicasting delay sensitive data achieving the required packet delivery probability with a correlation coefficient of 0.98. Furthermore, up to 5% throughput gains are observed using the proposed algorithm over

reference algorithms. Subsequently, practical network coding with pre-coding and interleaving is presented to improve the robustness of PNC against bursty packet losses. The proposed method is capable of increasing the throughput of a data multicast by up to 4.5% compared to classical PNC against packet loss bursts.

Rest of the proposed NC algorithms is content-aware and is optimized for both data and video multicast applications. A set of content and network aware NC algorithms are proposed to increase the performance of delay sensitive data and video multicast applications in packet lossy networks. These algorithms optimize the redundancy allocation for NC considering content properties as well as network status. In doing so, the probability of decoding an NC generation of a data multicast is improved by up to 25% compared to reference algorithms. As for video multicast applications, up to 5dB PSNR gains of the received video quality and 18% increase in the probability of continuous video playback are observed over state-of-the-art reference algorithms.

Furthermore, novel algorithms are designed to jointly optimize channel coding and NC by considering content properties and channel status to deliver data and video multicast services across wireless networks. A set of content and network-aware distributed algorithms is presented to efficiently multicast data and video by independently allocating cross-layer redundancies at each NC enabled network node in a wireless network. Up to 10× reduction in the average number of redundant bits transmitted by a network node for data multicast applications and up to 3.5 dB gains in the received video quality of a scalable video multicast applications are observed over state-of-the-art reference schemes.

In addition, the possibilities of optimizing the redundancy allocation for joint source and network coding are explored to increase the robustness of a video multicast. Up to 8dB PSNR gains are observed through simulation results when redundancies are applied as NC packet combinations compared to video redundancies applied at the application layer along with multipath multicast. In addition, it is observed that existing video coding techniques are not optimal for video multicast applications in PNC enabled networks. This observation encourages designing NC aware video coding algorithms to perform joint optimization of source-channel and network coding to increase the video quality of future high volume video multicast applications.

In conclusion, the work presented in this thesis proposes several novel NC algorithms to optimize the performance of multicast applications. Since video traffic occupies majority

of the total network traffic, special attention is paid to design content aware NC algorithms to efficiently utilize network resources to multicast video at a high quality. Simulation results of proposed solutions outperforming state-of-the-art reference techniques prove the capability of the proposed solutions to improve the performance of multicast applications.

7.3 Future Work

This section points out possible future work related to the work presented in this thesis.

7.3.1 Joint Source Channel and Network Coding for Efficient Video Multicast Applications

Chapter 6 of this thesis points out that existing video coding algorithms are not optimal for PNC enabled networks. This signifies the requirement of novel NC aware video coding techniques to allocate redundancies for joint source channel and network coding applications. By means of harmoniously optimizing source channel and network coding, network resources can be efficiently utilized to deliver high volume video multicast services at a satisfactory quality.

7.3.2 Efficient Data Encryption Models for Secure Multicast Applications

One of the advantages of PNC is that it is difficult to eavesdrop the transmitted content without collecting sufficient packet combinations to decode NC generations. Collecting sufficient packet combinations to decode NC generations is a challenge due to the capability of PNC to multicast data across diverse multiple paths. Utilizing the SCM, which is used in the algorithms proposed in Sections 3.2 and 3.4 of this thesis, already provides an additional encryption layer between source node and sink nodes. This aspect can be further explored to design strong encryption mechanisms to support secure data communication across PNC enabled network.

7.3.3 Performing Network Tomography in PNC Enabled Networks

Network tomography in PNC enabled networks is a challenge due to network nodes using randomly generated coefficients to perform PNC. However, it should be noted that the algorithm proposed in Section 3.2 of this thesis uses coefficient information to identify

which source packet combinations are re-combined to generate any NC packet combination in the network. Using this information, sink nodes can identify which packets have been combined across different paths it receives packets from. This property can be further exploited to improve existing techniques to perform network tomography in PNC enabled networks.

7.3.4 NC Based Distributed Storage Systems for Future Communication Networks

This thesis proposes NC algorithms to increase the throughput and robustness of data and video multicast applications by efficiently utilizing available bandwidth resources. However, the same concepts can be exploited to design NC based fault tolerant distributed storage systems to efficiently utilize storage resources to save data and video content. Furthermore, the proposed content aware NC algorithms to selectively transmit video across networks can be adopted to save high volume video content in a fault tolerant manner when available storage space is limited.

7.3.5 NC Based Content-Aware Fault-Tolerant Multiple-Cloud Storage Systems

NC based fault tolerant algorithms are capable of providing a reliable fault-tolerant multiple-cloud storage over non-NC based approaches [112]. However, existing NC based fault-tolerant multiple-cloud storage systems are content agnostic. Hence, they are not optimal for video storage. The content-aware algorithms proposed in this paper for efficient video multicast applications could be adopted to develop NC based content-aware fault-tolerant multiple-cloud storage systems. This will improve the fault tolerance capabilities of existing NC based multiple-cloud storage algorithms for multiple-cloud video storage applications.

7.3.6 Content-Aware Joint Congestion Control and Scheduling in Wireless Networks with Network Coding

NC enabled network nodes buffer NC packet combinations (as explained in Section 2.3). However, the performance of an NC enabled network may be reduced unless NC packets saved in the buffer of network nodes are updated in an intelligent manner [113]. In

response, several algorithms, including [113], propose joint congestion control and scheduling algorithms in NC enabled wireless networks to minimize the required buffer size without compromising the network performance. However, these algorithms are not content-aware. Therefore, the algorithms proposed in Chapter 4 can be developed to enhance the performance of existing algorithms to minimize the memory requirements of NC enabled nodes for video transmission applications.

7.3.7 Low Latency NC Schemes for Real-Time Video Communication and Gaming Applications

Several algorithms, including [114] are presented to minimize the encoding/decoding latency of NC by introducing fast random encoding parallelization techniques, which target to efficiently utilize multi-core processors. The performance of these algorithms for real-time video communication and online gaming applications can be improved by considering content properties and intelligently allocating processing resources such that high priority NC packets can be encoded/decoded with minimum delay.

7.3.8 Low Complexity Content-Aware Joint Channel and Network-Coding Algorithms for Wireless Sensor Networks

Wireless Sensor Networks (WSN) have a wide range of applications such as military surveillance, environment monitoring, target tracking, etc. Several algorithms are available in the literature to optimize the performance of such networks using NC techniques [115]. The performance of such algorithms can be improved by developing the algorithms proposed in Chapter 4 to design low complexity content-aware joint channel and network-coding algorithms.

7.3.9 Optimize NC Algorithms for Software Defined Networking

Software Defined Networking (SDN) abstracts the lower level functionalities of networks such that it is convenient to manage network services. However, existing NC algorithms are not designed considering the OpenFlow architecture of SDNs. On the other hand, abstraction of the functionalities in lower layers in a network and enabling to optimize network services can be leveraged to improve the performance of existing NC based

algorithms. Hence, novel NC based algorithms optimized for SDN can be designed to efficiently deliver data and video in future communication networks.

7.4 Summary

This chapter summarizes the main contributions of this thesis and presents several future research directions. In summary, this thesis presents novel PNC algorithms to increase the throughput and robustness of data and video multicast applications. By means of performing extensive simulations, it is proved that the proposed algorithms outperform existing state-of-the-art algorithms. Hence, it is envisaged that the use of the proposed algorithms will provide efficient solutions to cater high volume data and video multicast services in future communication networks. What is more, the research directions pointed out in this chapter will further improve the performance and usability of the proposed algorithms.

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