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ANALYSIS OF EFFICIENT WIRELESS MULTICAST RETRANSMISSION AND COOPERATIVE RELAYING TECHNIQUES BASED ON MULTIPLE CODED PACKETS

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Master of Science Thesis

Analysis of Efficient Wireless Multicast Retransmission and Cooperative Relaying Techniques based on Multiple Coded Packets

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Dissertação submetida à Coordenação do Programa de Pós-Graduação em Engenharia de Teleinformática, da Universidade Federal do Ceará, como requisito parcial para obtenção do grau de Mestre em Engenharia de Teleinformática.

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To my darling family.

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Resumo

Esta tese de mestrado aborda a aplicação de técnicas de cooperação para redes sem fio com o objetivo de melhorar a sua eficiência espectral. As técnicas cooperativas consideradas são, mais especificamente, baseadas em codificação de rede em um contexto celular multicast sem fio, bem como em esquemas de repetidores (*relaying*) em cenários com até dois saltos.

Este trabalho avalia o desempenho de diferentes algoritmos para transmissão em redes multicast sem fio considerando uma única célula. Uma técnica eficiente de retransmissão para serviços multicast baseado em codificação de rede com o uso de um buffer é proposto e comparado com outros esquemas tradicionais, como o caso sem retransmissão e com retransmissão simples, bem como o esquema de retransmissão com múltiplos pacotes codificados. É mostrado que o algoritmo proposto supera outras técnicas e a qualidade da transmissão multicast é melhorada.

Além disso, esta tese de mestrado avalia o impacto do erro de estimação de diferentes algoritmos através de um cancelamento imperfeito da auto-interferência, a qual surge no contexto da comunicação bi-direcional em cenários com *relays* em uma única célula. Uma técnica eficiente de transmissão para serviços de comunicação bi-direcional baseada em *relays* é comparada com outros esquemas. Foi demonstrado que um cancelamento adequado da auto-interferência é crucial para o uso mais eficiente dos recursos de rádio disponíveis.

Palavras-chave: Codificação de rede, técnicas de retransmissão, multicast sem fio, repetidores.

Abstract

This master's thesis tackles the application of cooperative techniques to wireless networks with the purpose of improving their spectral efficiency. The considered cooperative techniques are, more specifically, based on network coding in a wireless multicast cellular context, as well as on relaying schemes in two-hop scenarios.

This work evaluates the performance of different algorithms for wireless network multicast transmission in a single cell. An efficient retransmission technique for multicast services based on network coding with a buffer is proposed and compared to other traditional schemes, such as the case with no retransmission, a simple retransmission and retransmission with multiple coded packets scheme. It is shown that the proposed algorithm outperforms other techniques and quality of the multicast transmission is improved.

Moreover, this Master's thesis evaluates the impact of estimation error of different algorithms through an imperfect Self-Interference (SI) cancellation for Amplify-and-Forward (AF) bi-directional communication relaying in a single cell. An efficient transmission technique for bi-directional communication services based on Relay-Station (RS) is compared to other schemes. It was shown that an adequate cancellation of the SI is crucial for using more efficiently the available radio resources.

Keywords: Network coding, retransmission techniques, wireless multicast, relaying.

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List of Acronyms

3G	3rd Generation
3GPP	3rd Generation Partnership Project
AF	Amplify-and-Forward
AWGN	Additive White Gaussian Noise
BAT	Bi-directional Amplification of Throughput
BC	Broadcast
BS	Base-Station
DF	Decode-and-Forward
LTE	Long-Term Evolution
MAC	Medium Access Control Layer
MBMS	Multimedia Broadcast Multicast Service
MS	Mobile-Station
RF	Radio Frequency
RS	Relay-Station
SI	Self-Interference
SNR	Signal-to-Noise Ratio

UMTS Universal Mobile Telecommunications System

UTRAN UMTS Terrestrial Radio Access Network

Contents

Chapter 1

Introduction

1.1 Overview

In recent years the area of telecommunications has been the focus of many research and technological advances and, in this field, the area of wireless networks was the one that presented the most impressive improvements. By allowing connection to telecommunications systems, these networks have become a promising field with prospects of even greater future use.

Among the wireless technologies, the mobile telephone was the one that evolved the most and for which a great deal of applications and features have been developed. The mobile telephone systems offer the most advanced types of services and also support several types of traffic.

Advanced wireless systems such as the UMTS Terrestrial Radio Access Network (UTRAN) Long-Term Evolution (LTE)-advanced in 3rd Generation Partnership Project (3GPP) [1], have performance requirements in the physical layer that are difficult to reach with transceivers of conventional access technologies. In particular, in terms of spectral efficiency, a notable increase is expected when compared to 3rd Generation (3G) systems. In order to provide this increased spectral efficiency, effective radio resource management techniques must be implemented. In the case of multicast services, network coding can be employed to improve the overall throughput.

1.1.1 Wireless multicast

Wireless multicast [2, 3] is an important service with applications to file distribution, delay tolerant networks, home entertainment, and video conferencing. As more wireless networks are deployed on city-wide scales, and as mobile wireless devices continue to replace their fixed wired counterparts, this importance of wireless multicast will increase.

Through wireless multicast transmission [4] various MSs can receive the same information which is transmitted at the same radio resource. A resource can be understood in various ways such as a time-slot, a particular Radio Frequency (RF) or a spatial location. This form of transmission is very useful, for example, in the digital TV application, where several MSs of a particular cell generate a demand for data on the same channel. In the case of UTRAN and its evolution, the multicast services are specified by the Multimedia Broadcast Multicast Service (MBMS) standard [5], which defines resource allocation and transmission procedures specific for multicasting.

One particular bottleneck of wireless multicast services is the retransmission of erroneously received packets. Since a single resource is used, the retransmission of a packet occupies the resource and those MSs who had received it correctly have to wait until a new packet is transmitted. Some efficient retransmission algorithms for error-tolerant multicast services have been proposed in [6].

1.1.2 Network coding

In order to improve the multicast performance, we propose to employ network coding [7–10], which is a relatively new area of networking, in which data is manipulated inside the network to increase the received data throughput, reduce delay, and improve robustness. This field has recently found commercial applications in content distribution, peer-to-peer design, and enabling high-throughput wireless networks.

More generally, network coding combines packets before transmitting them along the network nodes and has two important benefits relevant to multicast routing. First, it decreases the number of transmissions necessary to route packets to multiple receivers for both multi-hop and single-hop routing. Second, it reduces the need for coordination among network nodes in multi-hop routing.

The combination of packets can be done by a simple logical operator such as XOR [9]. There is also the random network coding technique, which was shown reach a good performance in recent studies [11–13].

This network coding method allows for a natural and efficient means of loss recovery in face of low-quality wireless links and provides for economical path diversity, which is particularly important for multicast traffic in the unstable and lossy environments characteristic of wireless networks. In this work, the network coding studies we will consider only a single-hop scenario.

Network coding has some flexibility in terms of how to select which packets to combine, allowing to properly exploit the diversity of the multiple radio links. By mixing packets, network coding is able to reduce the number of transmissions necessary to convey packets to multiple receivers, which can lead to a large increase in performance for multicast traffic. Thus, network coding has the advantage of sending different packets in a single resource, which are coded into a single packet.

1.1.3 Cooperative relaying

In wireless cellular networks, the deployment of fixed RSs has been considered as an alternative for improving not only cell coverage but also transmission efficiency. Recent studies have focused on the analysis of adequate deployment scenarios, taking into account aspects such as the cost/efficiency trade-off of RSs as well as resource allocation issues [14–16].

There are two main types of relaying techniques: the AF and the Decode-and-Forward (DF). The first one only amplifies the signal received by the RSs before forwarding it, while the second one decodes these data symbols.

The main advantage of the AF relaying technique is the benefit of lower implementation complexity than DF techniques. In the AF scheme, the received estimates of the data symbols are further transmitted without any decoding attempts at the RS, which causes a save radio resources and reaches a good performance in certain scenarios. On the other hand, this technique presents an undesirable noise amplification effect.

1.2 Methodology

In complex wireless communication systems, the computer simulation appears as an effective analysis tool for evaluating system performance and proposing improvements in its major functions and architectural aspects. However, this approach requires the selection of models capable of providing detailed information about the system, as well as the proper selection of software tools enabling the effective use of computer resources.

1.3 Thesis organisation

The main contribution of this thesis is the proposal of two efficient multicast retransmission algorithms based on network coding, one with multiple coded packets and another that additionally assumes a buffer that stores network coded packets at the receivers, whose performance is analyzed and compared to that of other retransmission schemes.

Moreover, this thesis analyzes the impact of channel estimation error on the performance of a solution of a bi-directional communication via RS. This Master's thesis is divided into five chapters. The contents of each of the next four chapters are stated below.

Chapter 2: describes the system model and presents a detailed description and analysis of the considered algorithms of multicast retransmission techniques based on multiple coded packets;

Chapter 3: describes the general system model and is found the derivation of the individual multi-antenna relaying for bi-directional communication algorithms;

Chapter 4: shows the performance analysis of numerical results;

Chapter 5: the final conclusions and perspectives are drawn.

Chapter 2

Multicast retransmission techniques based on multiple coded packets

2.1 Introduction

In [17], network coding techniques are proposed to reduce the number of broadcast/multicast transmissions from one sender to multiple receivers, thus increasing the bandwidth efficiency of reliable multicast in a wireless network. The main idea of employing network coding to multicast retransmissions is to allow the sender to combine and retransmit different lost packets from different receivers in a way that multiple receivers are able to recover their own lost packets with one transmission by the sender.

The main contribution of this chapter is the proposal of two efficient multicast retransmission algorithms based on network coding, one with multiple coded packets and another that additionally assumes a buffer that stores network coded packets at the receivers, whose performance is analyzed and compared to that of other retransmission schemes.

This chapter is divided as follows: section 2.2 describes the scenario and the multicast quality-of-service metric and the section 2.3 presents a detailed description and analysis of the considered algorithms.

2.2 Considered scenario and measures

This section describes the scenario and the multicast quality-of-service metrics. The system assumes a uniform spatial distribution of users within a single cell. This group of users are part of a multicast service where the transmission occurs in the downlink through the same radio resource and all users see a same average interference.

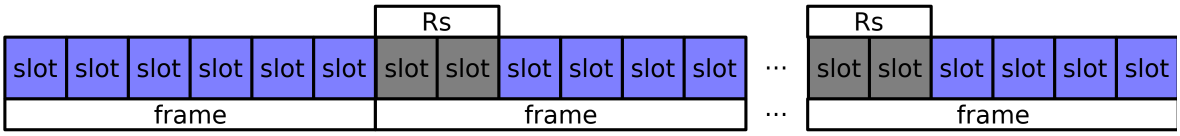


Figure 2.1: The temporal structure of the system model.

Figure 3.1 represents the temporal structure of the system model. The time was divided in frames and each frame in slots. It is assumed that a packet is transmitted at each slot. Within each frame a maximum number of retransmission slots per frame (R_s) is reserved for the retransmission of packets, with retransmissions starting to occur after the first frame. It is also assumed that at the beginning of each frame the base station has gathered the feedback from the users regarding which packets were not received correctly in the previous frame.

Another parameter is the retransmission criterion. We assume that a retransmission of a packet is allowed to occur only when a certain percentage of the users have not received it correctly, as it is done for example in [6]. This parameter is intended to reduce the delay of the system, since it is not worth utilizing a retransmission slot for only a few users. Let U_{T} denote the total number of users in the system and U_{R} the number of users who have not received a specific packet. The retransmission criterion for a given packet is satisfied when $\alpha > \alpha_{\mathrm{th}}$, where α is given by

$$\alpha = \frac{U_{\mathrm{R}}}{U_{\mathrm{T}}}, \tag{2.1}$$

and α_{th} is the threshold required for retransmission. Note that this threshold allows to emulate the error tolerance of the service. For $\alpha_{\mathrm{th}} = 0$, it means that the service does not tolerate any errors, otherwise, if $\alpha_{\mathrm{th}} > 0$, the service tolerates the loss of some packets.

All graphics in the results section have curves that quantify the amount of successfully received packets of the algorithms in terms of a specific parameter. We consider two different measurements, with the first one indicating the multicast quality-of-service in terms of an efficiency η, which is given by

$$\eta = \frac{L_{\mathrm{success}}}{U_{\mathrm{T}} S_{\mathrm{tx}}}, \tag{2.2}$$

where L_{success} corresponds to the total number of correctly received packets of all users, U_{T} is the total number of users in the system, and S_{tx} is the effective total number of transmitted slots used.

The second metric also indicates the multicast quality-of-service, but in terms of the normalized system delay d, which is given by

$$d = \frac{S_{\text{rtx}}}{S_{\text{tx}}}, \tag{2.3}$$

where S_{rtx} is the effective total number of retransmitted slots used.

2.3 Retransmission algorithms

This section describes the four different algorithms considered by the simulation analysis, which are namely: no retransmission (NRTX), simple retransmission (SRTX), network coded retransmission with the combination of multiple packets (NCRTXp, where p represents the number of combined packets) and network coded retransmission with 2 packets and buffering at the receivers (NCRTX2B). The algorithms described below differ in how they choose or combine the packets for retransmission.

2.3.1 No retransmission

The first algorithm is the simplest of all and has transmission with no retransmission. Thus, different packets are transmitted at each slot, resulting in a total number of packets equivalent to the number of frames times the number of slots per frame. Since there are no retransmissions, the calculation of d always has the value null because it simply takes into account all slots, i.e., the number of transmitted packets is the same as the total number of simulated slots.

2.3.2 Simple retransmission

The second algorithm corresponds to a simple retransmission. Thus, beginning from the second frame, a maximum number of retransmission slots per frame is reserved. Based on the feedback from the users, the base station computes for each transmitted packet how many users received it correctly.

Let $U_{i,j}$ denote the number of users that correctly received packet i in frame j. The packets are ordered according to the increasing value of $U_{i,j}$, i.e., priority is given to those packets that are received by the smallest amount of users. In frame $j+1$ the packets with the highest priority are retransmitted within the reserved retransmission slots, provided that a minimum tolerated percentage of users who have not received this packet is reached.

There is an exception when there are less packets to retransmit than the reserved amount of retransmission slots per frame. In this case, in order to avoid idle slots, these free retransmission slots are used for transmitting new packets.

2.3.3 Network coded retransmission with multiple coded packets

The third algorithm applies network coding to retransmit multiple packages in a single coded packet. Similarly to the previous algorithm, the same steps are done, but in frame $j+1$ coded packets are sent within the slots reserved for retransmission. The packets to be retransmitted are combined in pairs, trios, quartets and quintets. Let n denote the number of packets encoded. Within each frame a maximum of nR_s packets can therefore be retransmitted.

Let L_q denote the total number of packets queued for retransmission, where the packets are ordered according to the same priority scheme of the previous algorithm, and L_j the number of packets to be retransmitted within frame j. For this algorithm the following relationship holds: $L_j = \min\{L_q, nR_s\}$. The L_j packets with the highest priority are selected and they are combined by successively picking the head and tail of this queue of selected packets.

When L_j is not a multiple of n, the intermediary packets will be encoded and retransmitted, i.e., a coded packet will contain less than n packets. The actual packet encoding can be done by a simple logical operator such as XOR [9]. Note that the same exception of the simple retransmission algorithm with regard to free retransmission slots also holds.

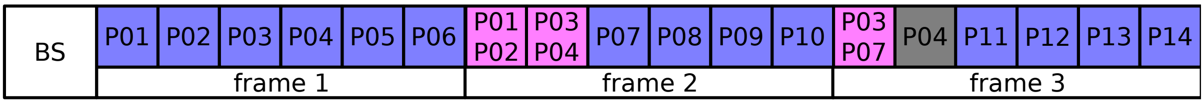
The packet combination is made as proposed because, in order to achieve gains with network coding, the user receiving the coded packets needs to already have correctly received at least $n-1$ of the packets in a previous transmission, so that it can decode the other one. According to the priority scheme, the packets at the head of the queue have been received by the least amount of users, so that if they were to be combined in order, the probability of a user having previously received at least $n-1$ of the packets would be quite low.

The proposed combination scheme is therefore a low-complexity algorithm that aims at increasing this probability. The packet combination proposed has a gain compared with a random packet combination.

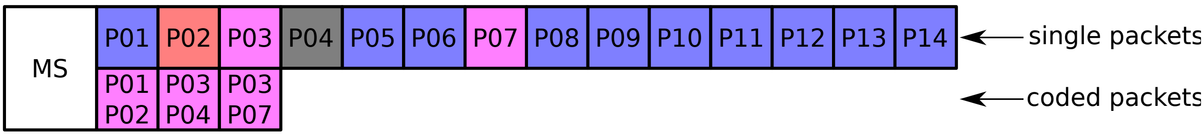
2.3.4 Network coded retransmission with buffering at the receivers

The last algorithm also has retransmission for network coding and in encoded form, but uses a buffer to store packets encoded in the mobile stations. This is the proposed algorithm that aims at reducing the total number of slots used for retransmission. Similarly to the NCRTX2 algorithm, the same steps are done, but in frame $j+1$ the mobile stations store coded packets sent within the slots reserved for retransmission.

Upon receiving the retransmission of a single packet, the mobile stations will check the buffer of coded packets to detect whether some additional packet can be decoded. The mobile stations only need to store coded packets containing at least one packet that has not yet been decoded.



(a) Time structure of BS transmission.



(b) MS buffer.

Figure 2.2: Example of NCRTX2B algorithm.

Figure 2.2 represents an example of the NCRTX2B algorithm. In this example, each frame was divided in six slots where two are reserved for retransmission. Initially the base station (BS) transmits the first six packets in the first frame, but the mobile station (MS) receives only the packets P01, P05 and P06.

In the second frame, two coded packets are retransmitted: the first containing the packets P01 and P02, and the second containing P03 and P04. Assuming MS receives both, MS stores these packets and the algorithm, through the first coded packet, can decode P02 in the same way as the NCRTX2 algorithm. The BS transmits the following four packets next, but MS receives only packets P08, P09 and P10.

Finally, in the third frame, packets P03 and P07 are retransmitted in encoded form, and packet P04 is retransmitted unencoded, since there are no further packets to be retransmitted at the moment. Assuming the MS receives both, it stores the coded packet and the algorithm, through the second coded packet and P04, can decode P03. Moreover, through the third coded packet and P03, the algorithm can decode P07.

Chapter 3

Impact of imperfect self-interference cancellation for bi-directional relaying

3.1 Introduction

Assuming the availability of RSs within a cell for allowing two-hop communication between pairs of MSs and Base-Station (BS), the focus of this chapter lies on how to improve the efficiency of both link directions, more specifically on how to evaluate the impact of estimation error for AF bi-directional communication through an RS.

Previous works have mainly approached cases where there is a single pair of MS/BS communicating through an RS [18–20], or more recently with multiple pairs sharing a same RS [21, 22]. In this chapter we focus on a scenario considering a single BS, with homogeneous deployment of RSs within the cell, allowing two-hop communication between the MSs and the BS.

This scenario has been approached for example in [23, 24], but considering diversity techniques. Here we analyze some possible approaches with the purpose of improving the transmission efficiency, such as the bi-directional AF, which employs Self-Interference (SI) cancellation to reduce the required amount of radio resources. In this sense, the impact of imperfect SI cancellation is analyzed, which is a critical practical problem that might degrade performance.

The main contribution of this chapter is to analyze the impact of imperfect SI cancellation for bi-directional relaying. This chapter is organized as follows: section 3.2 describes the general system scenario and the sum rate quality-of-service metric, and the derivation and a detailed description and analysis of the considered individual relaying algorithms are found in section 3.3.

3.2 Considered general system scenario model and assumptions

This section describes the general system scenario and the sum rate quality-of-service metric. The general system scenario assumes a uniform spatial distribution of MSs and a linear distribution of RSs around a circumference within a single cell. There is only one BS in the center of this cell and the RSs are at the same distance from the BS. This group of nodes is part of a bi-directional communication service where the transmission occurs in both downlink and uplink.

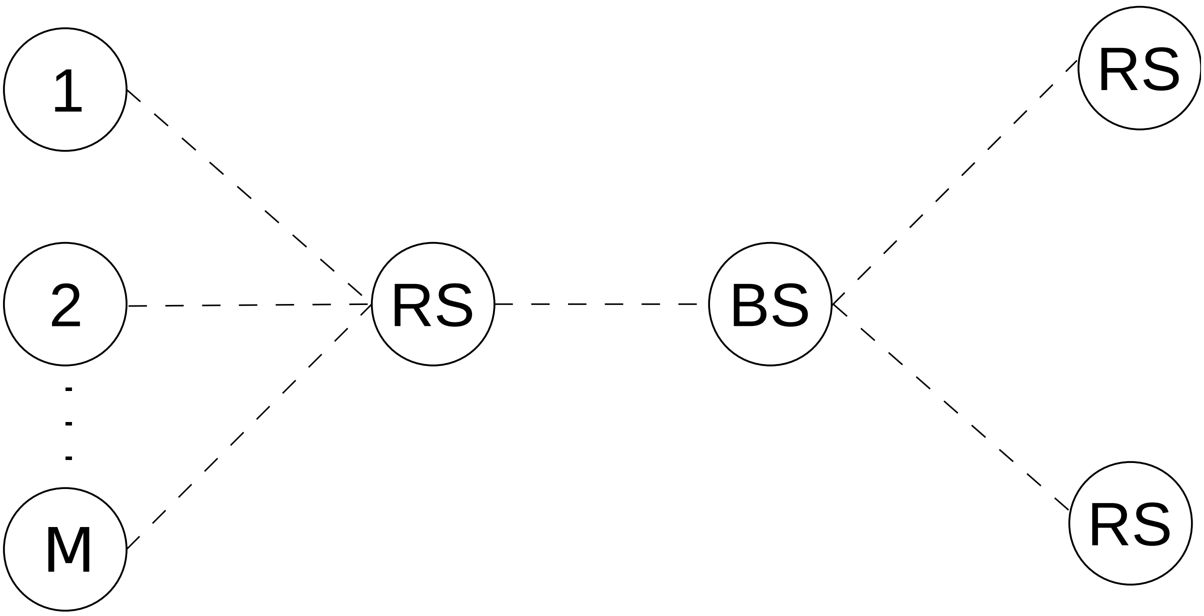


Figure 3.1: The system model.

The general system model is depicted in Fig. 3.1. There are N MSs, some RS and a single BS, all with single-antenna capabilities. It is assumed that the MSs cannot connect directly to the BS, as in a situation of strong shadow fading, thus being restricted to a two-hop communication through the RSs. It is further assumed that each MS has data to send to the BS, as well as the BS has data to send to each MSs.

The AF relaying technique, in which the received estimates of the data symbols are further transmitted without any decoding attempts at the RS, is considered by all algorithms in the upcoming sections. In spite of the undesirable noise amplification effect that AF presents, it has been shown to reach good performance in certain scenarios, with the benefit of lower implementation complexity than DF techniques.

In order to have a common framework for comparing the algorithms, a single frequency resource is considered, with orthogonal transmissions being separated in the temporal or spatial domains. Let C denote the sum rate metric of the transmission between MSs and BS in both directions:

$$C = \frac{1}{2N} \sum_{i=1}^{N} \left\{ \log_2(1+\gamma_{i,b}) + \log_2(1+\gamma_{b,i}) \right\}, \tag{3.1}$$

where N is the number of MSs, $\gamma_{i,b}$ is the equivalent Signal-to-Noise Ratio (SNR) between the i-th MS and the BS, and $\gamma_{b,i}$ the equivalent SNR between the BS and MS i.

The channel between each pair of nodes is assumed to be composed of a zero mean circularly symmetric complex Gaussian random variable with unit variance $z \in \mathbb{C}$ and a path-loss component $L \in \mathbb{R}$, i.e., $h_{i,j} = \sqrt{L_{i,j}} z_{i,j}$. Let σ_m^2, σ_r^2, and σ_b^2 denote the Additive White Gaussian Noise (AWGN) variance of the MSs, RSs, and BS, respectively, with P_m, P_r, and P_b denoting the transmit power constraints of the corresponding nodes.

It should also be mentioned that flat fading is assumed, such that there is no inter-symbol interference, and the channel is considered to remain approximately the same for a certain period of time, during which several symbols can be transmitted subject to different noise samples. When calculating the average SNR for a given channel realization we thus assume that the channel is constant and symbol and noise are random variables, from which the expected values can be taken.

3.3 Relaying algorithms

In this section the relaying algorithms are described. First, we consider a simple one-way AF relaying case, where each transmission occupies a time slot. Next comes the two-way AF relaying, which accepts symbol superposition. All algorithms consider this scheme, where each bi-directional transmission occupies a time slot.

3.3.1 One-way amplify-and-forward relaying

This simple case corresponds to the sequential application of one-way AF relaying to each pair MS-BS. It requires a total amount of $4N$ time slots to complete the information exchange among all pairs of nodes. The transmission from one node i to another node j can be represented by:

$$y_r = \sqrt{P_i} h_{i,r} s_{i,j} + n_r \,, \tag{3.2}$$

$$y_j = \beta_{r,j} h_{r,j} y_r + n_j \,. \tag{3.3}$$

where y_r represents the symbol received at the RS after one time slot and y_j the symbol received at node j after the following time slot, $s_{i,j}$ is the symbol sent from i and intended to j, n represents the noise at a certain node, $\beta_{r,j}$ is a positive real scalar associated to the relay power constraint. This scheme is roughly illustrated in Fig. 3.2, when considering the one-way transmission from one node to the other. Note that the way back for this MS will take two further time slots.

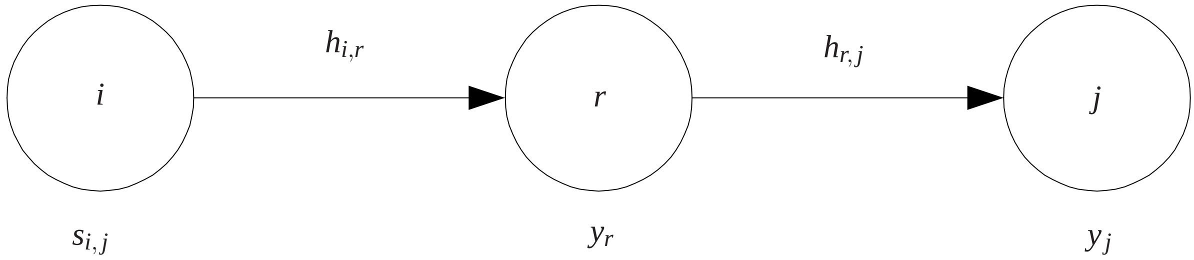


Figure 3.2: Simplified diagram of one-way relaying, representing a generic sequence of two slots.

Note that the RS has to forward the received y_r symbol while satisfying its transmit power constraint, i.e., $\mathrm{E}\{|\beta_{r,j} y_r|^2\} = P_r$. When assuming that the symbols and the noise are independent random variables and that $\mathrm{E}\{|s|^2\} = \sigma_s^2 = 1$, it follows that:

$$\beta_{r,j} = \sqrt{\frac{P_r}{P_u |h_{i,r}|^2 + \sigma_r^2}} \,. \tag{3.4}$$

The signal received in j can be expanded to

$$y_j = \sqrt{P_i} \beta_{r,j} h_{r,j} h_{i,r} s_{i,j} + \beta_{r,j} h_{r,j} n_r + n_j \,, \tag{3.5}$$

from which the SNR $\gamma_{i,j}$ can be written as

$$\gamma_{i,j} = \frac{P_i \beta_{r,j}^2 |h_{i,r} h_{r,j}|^2}{\beta_{r,j}^2 |h_{r,j}|^2 \sigma_r^2 + \sigma_j^2} \,. \tag{3.6}$$

The sum rate of this relaying technique, when applying (3.1), becomes

$$C_{1\text{W}} = \frac{1}{4N}\sum_{i=1}^{N}\left\{\log_2(1+\gamma_{i,b}) + \log_2(1+\gamma_{b,i})\right\}, \tag{3.7}$$

with the SNR γ given by (3.6), when replacing the corresponding indexes.

3.3.2 Two-way amplify-and-forward relaying

The simplified diagram of two-way AF relaying scheme for a given MS is depicted in Fig. 3.3.

The two-way AF relaying scheme divides the transmission into two phases: the Medium Access Control Layer (MAC) phase, where the MS and the BS transmit simultaneously to the RS, and the Broadcast (BC) phase, where the RS broadcasts the combined signal to both MS and BS. When this procedure is sequentially applied to all MSs, this results in a total of $2N$ required time slots, which corresponds to half the number of resources demanded by one-way relaying.

This superposition-based AF scheme is also known as Bi-directional Amplification of Throughput (BAT) relaying [18]. When the RS transmits the combined symbols, each receiving node should be capable of subtracting its SI. This principle is similar to the DF scheme employing network coding [22, 25], but instead of combining the decoded packets, the actual symbols are superimposed.

The system expressions for the MAC and BC phases of a given MS i are as follows:

$$\text{MAC}: y_r = \sqrt{P_m}h_{i,r}s_{i,b} + \sqrt{P_b}h_{b,r}s_{b,i} + n_r\,, \tag{3.8}$$

$$\text{BC}: \begin{cases} y_i = \beta_r h_{i,r} y_r + n_m\,, \\ y_b = \beta_r h_{b,r} y_r + n_b\,. \end{cases} \tag{3.9}$$

For the BC phase, the RS needs to constrain the transmit power of the combined symbol, such that $\mathrm{E}\{|\beta_r y_r|^2\} = P_r$, which leads to

$$\beta_r = \sqrt{\frac{P_r}{P_m|h_{i,r}|^2 + P_b|h_{b,r}|^2 + \sigma_r^2}}\,. \tag{3.10}$$

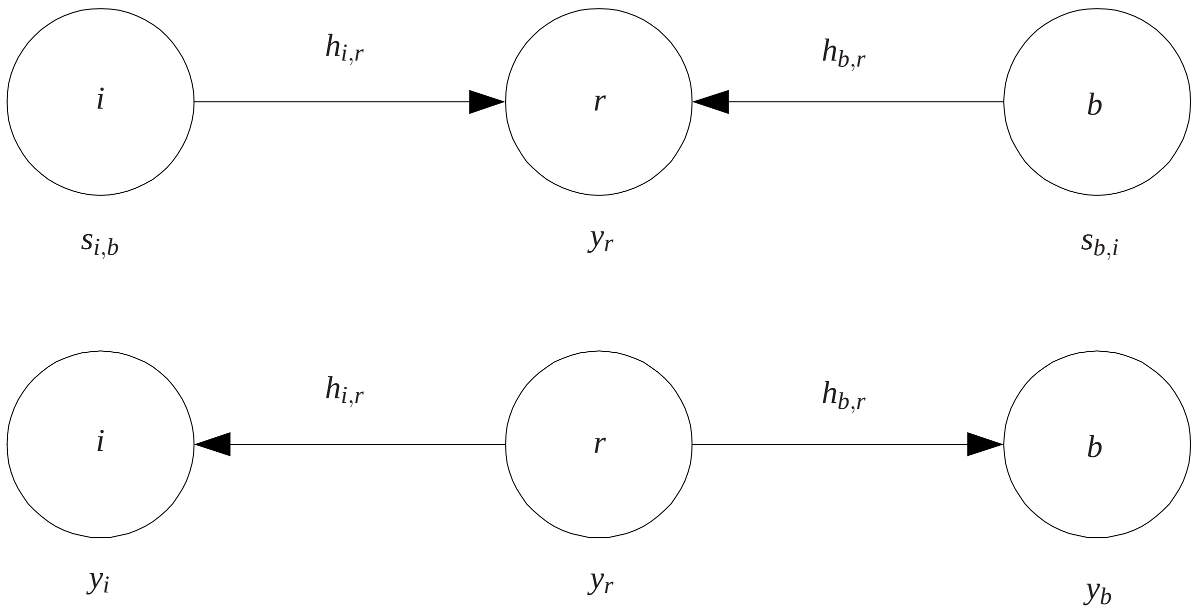


Figure 3.3: Simplified diagram of two-way relaying for a given MS.

Let us expand the expression for the signal received at the MS:

$$y_i = \sqrt{P_m}\beta_r h_{i,r}^2 s_{i,b} + \sqrt{P_b}\beta_r h_{i,r} h_{b,r} s_{b,i} + \beta_r h_{i,r} n_r + n_m \, . \tag{3.11}$$

The first summand corresponds to the SI which should be removed at the receiver, while the second summand represents the actual signal intended to the MS. In order to cancel the SI the following operation should be performed:

$$\hat{y}_i = y_i - \sqrt{P_m}\hat{\beta}_r h_{i,r}^2 s_{i,b} \, , \tag{3.12}$$

where $\hat{y}$ represents the estimated symbol. Note that the only parameter that is not readily available at the receiver is β_r, which must therefore be estimated as $\hat{\beta}_r$, e.g., through a previous signaling from the RS to all MSs. The SNR corresponding to the equivalent transmission from the BS to the MS can be written as:

$$\gamma_{b,i} = \frac{P_b \beta_r^2 |h_{i,r} h_{b,r}|^2}{P_m(\beta_r - \hat{\beta}_r)^2 |h_{i,r}^2|^2 + \beta_r^2 |h_{i,r}|^2 \sigma_r^2 + \sigma_m^2} \, . \tag{3.13}$$

The other SNR $\gamma_{i,b}$ can be obtained analogously:

$$\gamma_{i,b} = \frac{P_m \beta_r^2 |h_{i,r} h_{b,r}|^2}{P_b(\beta_r - \hat{\beta}_r)^2 |h_{b,r}^2|^2 + \beta_r^2 |h_{b,r}|^2 \sigma_r^2 + \sigma_b^2}. \tag{3.14}$$

Thus, the sum rate (C) is given by (3.1).

Algorithm 1

The first algorithm is the simplest of all and has an estimation of $\hat{\beta}_r$ which does not take into account both shadowing and fast fading of the channels. So, only the path-loss is used to calculate both $\hat{h}_{i,r}$ and $\hat{h}_{b,r}$ in $\hat{\beta}_r$ as follows:

$$\hat{\beta}_r = \sqrt{\frac{P_r}{P_m|\hat{h}_{i,r}|^2 + P_b|\hat{h}_{b,r}|^2 + \sigma_r^2}}. \tag{3.15}$$

Algorithm 2

The second algorithm corresponds to different ways of estimating $\hat{\beta}_r$ according to the calculated γ. In the case of $\gamma_{b,i}$, as it represents the SNR to the equivalent transmission from the BS to the MS, the estimated $\hat{\beta}_r$ does not take into account both shadowing and fast fading only of the channel $\hat{h}_{b,r}$. This means that the MS knows exactly the channel $h_{i,r}$, but estimated with some error the channel $\hat{h}_{b,r}$. Thus, in this case, only the path-loss is used to calculate $\hat{h}_{b,r}$ in $\hat{\beta}_r$ as follows:

$$\hat{\beta}_r = \sqrt{\frac{P_r}{P_m|h_{i,r}|^2 + P_b|\hat{h}_{b,r}|^2 + \sigma_r^2}}. \tag{3.16}$$

The other $\hat{\beta}_r$ used to calculate SNR $\gamma_{i,b}$ can be obtained analogously:

$$\hat{\beta}_r = \sqrt{\frac{P_r}{P_m|\hat{h}_{i,r}|^2 + P_b|h_{b,r}|^2 + \sigma_r^2}}. \tag{3.17}$$

Algorithm 3

The third and last algorithm also considers different ways of estimating $\hat{\beta}_r$ according to the calculated γ, but now the estimated $\hat{\beta}_r$ does not take into account only the fast fading of the estimated channel. Similarly to the previous algorithm, the same steps are done, but both path-loss and shadowing are used to calculate the estimated channel in $\hat{\beta}_r$ according to (3.16) and (3.17).

In addition to the analysis of the performance of the considered algorithms, in section 4.2 we have also established two bounding curves. The inferior one considers $\hat{\beta}_r = 0$ whereas the superior one considers $\hat{\beta}_r = \beta_r$.

Chapter 4

Results

4.1 Multicast retransmission techniques based on multiple coded packets

This section describes the simulation model of a multicast retransmission techniques based on multiple coded packets and a performance analysis and its numerical results are shown.

4.1.1 Simulation model

In the simulation model, all users have a different signal-to-interference-plus-noise-ratio (SINR) that depends on the additive white Gaussian noise (AWGN), the average interference power and the received signal power perceived by the users. The reception power depends on the transmission power, path loss, shadowing and gains of transmitting and receiving antennas.

The shadowing or slow fading can be caused when a large obstruction such as a hill or large building obscures the main signal path between the base station and the user. The amplitude change caused by shadowing was modeled using a log-normal distribution. The path loss is mainly a function of distance between the user and the base station. It was modeled according to [26] as follows:

$$PL = 120.9 + 37.6\log(dist)[dB], \tag{4.1}$$

where PL is the path loss and $dist$ the distance between user and base station.

In order to estimate the throughput achieved by the users, the SINR needs to be mapped to a certain packet error probability. We assume a simple mapping procedure consisting of two SINR thresholds, which determine the range of the packet error probability. The values associated to this mapping are modeled according to the following equation:

$$PEP = \begin{cases} 1\%, & \gamma > 30 \text{ dB}, \\ 10\%, & 0 \text{ dB} < \gamma \leq 30 \text{ dB}, \\ 99\%, & \gamma \leq 0 \text{ dB}, \end{cases} \tag{4.2}$$

where PEP is the packet error probability and γ is the SINR value to be mapped.

Note that these values roughly characterize a typical scenario. They have been inspired on block error rate link-level results of systems such as HSDPA and LTE [1]. Even though the absolute results may change significantly depending on this mapping, it is expected that the relative behavior among the algorithms remains approximately the same.

4.1.2 Numerical results and analysis

Table 4.1: *List of considered system simulation parameters for the multicast scenario.*

Parameter	Value	Unit
Cell radius	1	km
Number of users (U_T)	20	-
Transmission power (P_{tx})	40	dBm
Transmission gain	4	dBi
Reception gain	2	dBi
Interference power	25	dBm
Shadowing standard deviation	6	-
Noise power	-110	dBm
Simulated frames	20	-
Slots per frame	10	-
Retransmission slots (R_s)	2	-
Retransmission threshold (α_{th})	5	%
Iterations	10,000	-

A simulation tool was built to evaluate several different scenarios and algorithms concerning the provision of multicast services. The standard simulation parameters can be found in Table 4.2. In order to quantify the performance of the considered algorithms, the following key parameters have been chosen for analyzing their behavior:

i. Number of users within the cell;

ii. Total number of simulated frames;

iii. Power transmitted by the base station;

iv. Number of reserved retransmission slots per frame;

v. Number of slots per frame;

vi. Retransmission threshold (α_{th}).

The first parameter was chosen in order to analyze the impact of an increased load on the multicast system. The variation of the second parameter refers to the convergence of the simulation results. The third parameter has an impact on the radio link quality. Finally, the variation of the last parameters provide some insights on how to properly adjust these aspects of the transmission protocol.

Note that the NRTX algorithm, which has no retransmissions and is shown for comparison purposes, corresponds to a worst-case in terms of the efficiency (η) and as a best-case in terms of the normalized system delay (d). The delay of this algorithm is always null regardless of the variation of all key parameters, since the efficiency metric converges to the average packet success probability, which depends on the mapping given by (4.2). The only exception is the variation of the transmission power parameter, which has a direct impact on the perceived SINR.

The NCRTX2B algorithm, which has retransmission with a buffer and is the proposed algorithm, always has a better performance, or at least the same, compared to NCRTX2. This happens because the same steps are done, except the storage of coded packets in NCRTX2B. Thus, performance can be improved with the decoding of these packets.

Fig. 4.1 represents the graphic of the efficiency as a function of the number of users. For the SRTX algorithm, the higher the system load the lower the efficiency. Since the number of retransmission slots is fixed, the increase in the number of users ends up overloading the system, i.e., there are not enough resources to retransmit all packets.

With regard to the NCRTX2B algorithm, its efficiency is greater than the SRTX algorithm for more than 3 users. In return, he is surpassed by the algorithms with multiple packets encoded for more than 10 users. This is because, in the scheme of network coding with a buffer, the decoding of packets requires that the users should already know at least one of the packets in order to decode the other. The NCRTX2B algorithm highlights NCRTX2 for few users, in this way, these algorithms converge for many users in the system.

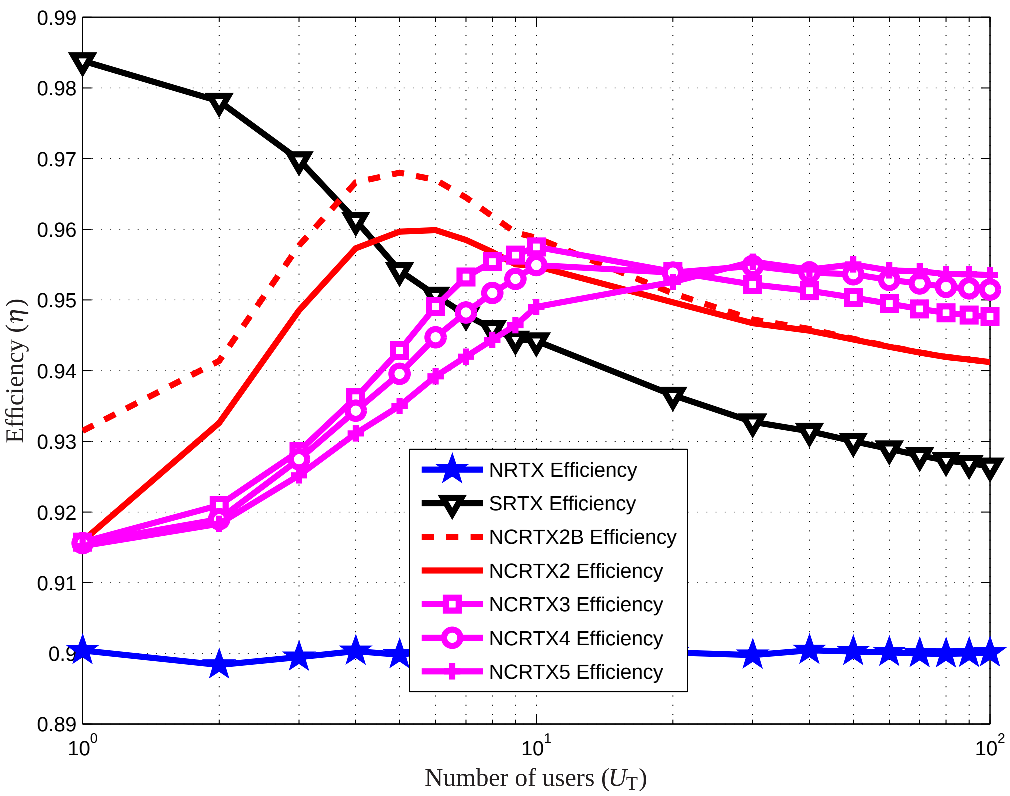


Figure 4.1: Impact of the number of users (U_T) on the efficiency (η).

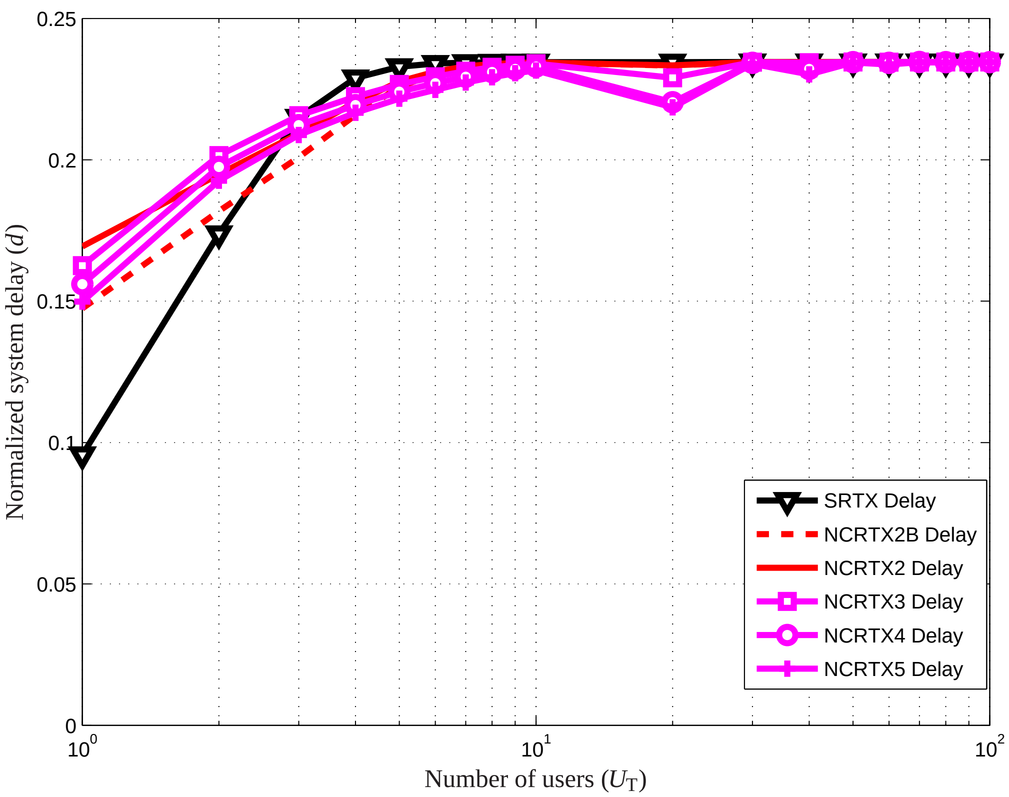


Figure 4.2: Impact of the number of users (U_T) on the normalized system delay (d).

Also, the algorithms with multiple packets encoded can retransmit more different packets according to the need of the multicast group. Moreover, the probability that the same user does not have any of the coded packets is very high for less than 4 users in the system. As soon as the number of users is enough to make up for this coding loss it follows the same behavior of the SRTX algorithm, i.e., efficiency decreases with more users.

Fig. 4.2 represents the graphic of the normalized system delay as a function of the number of users. Similarly to the previous metric, the higher the system load the greater the normalized system delay for all algorithms, since the normalized system delay metric is also sensitive to the number of retransmitted packets. The algorithms also present a similar behavior, increasing much more with the addition of the first few users, due to the same previously discussed reasons.

The impact of the number of frames on the efficiency and normalized system delay of the algorithms is shown in Figs. 4.3 and 4.4, respectively. When there is only one frame, all algorithms have the same efficiency and the value of normalized system delay is null, since no retransmissions take place. Both efficiency and normalized system delay increase with the number of frames.

For large numbers of frames the impact of the first frame without retransmission tends to be minimized, and therefore a convergence to average values is perceived for each algorithm and metric. The relative performance among the algorithms is the same as that of Figs. 4.1 and 4.2 for the case with 20 users, due to the previously explained reasons. Note that as convergence is achieved, the advantage of network coding schemes over the SRTX algorithm becomes more evident.

Figs. 4.5 and 4.6 show the impact of the transmission power. Since the interference power is constant at 25 dBm, increasing the transmit power is always beneficial for the users. The achieved results of the efficiency are directly related to (4.2). Regarding the normalized system delay metric, the algorithms with multiple coded packets present better performances.

Figs. 4.7 and 4.8 illustrate the efficiency and normalized system delay as a function of the maximum number of reserved retransmission slots per frame. Note that R_s has been simulated up to the number of slots per frame. So, it is possible to have more slots used for retransmission than for the actual transmission, i.e., in this case the delay is dramatically increased.

It can be seen that the efficiency and normalized system delay of the SRTX algorithm increases linearly up to four R_s while other algorithms up to two R_s. As for the efficiency of the algorithms, it converges with the number of Rs. This means that this number of Rs is already

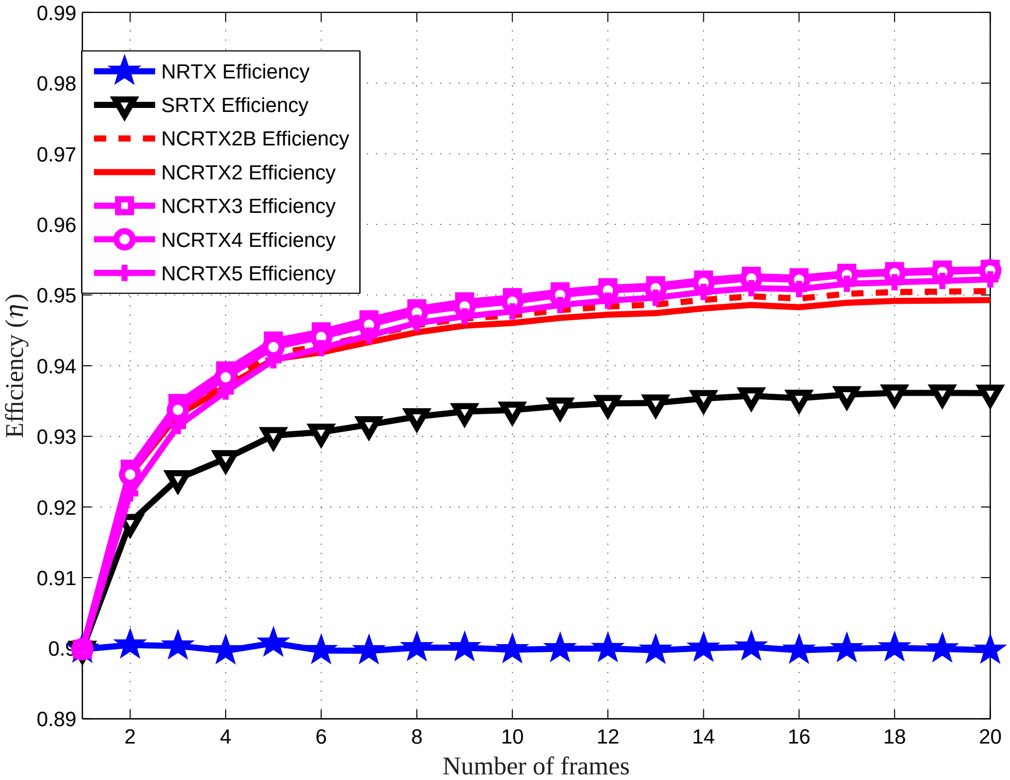


Figure 4.3: Impact of the number of frames on the efficiency (η).

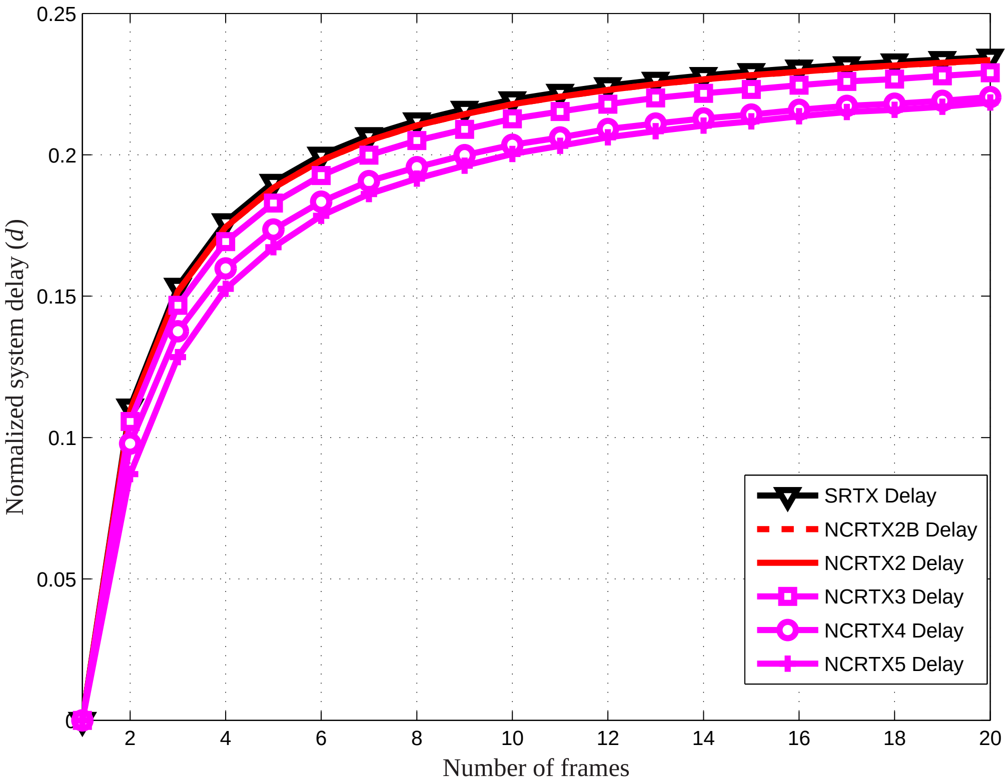


Figure 4.4: Impact of the number of frames on the normalized system delay (d).

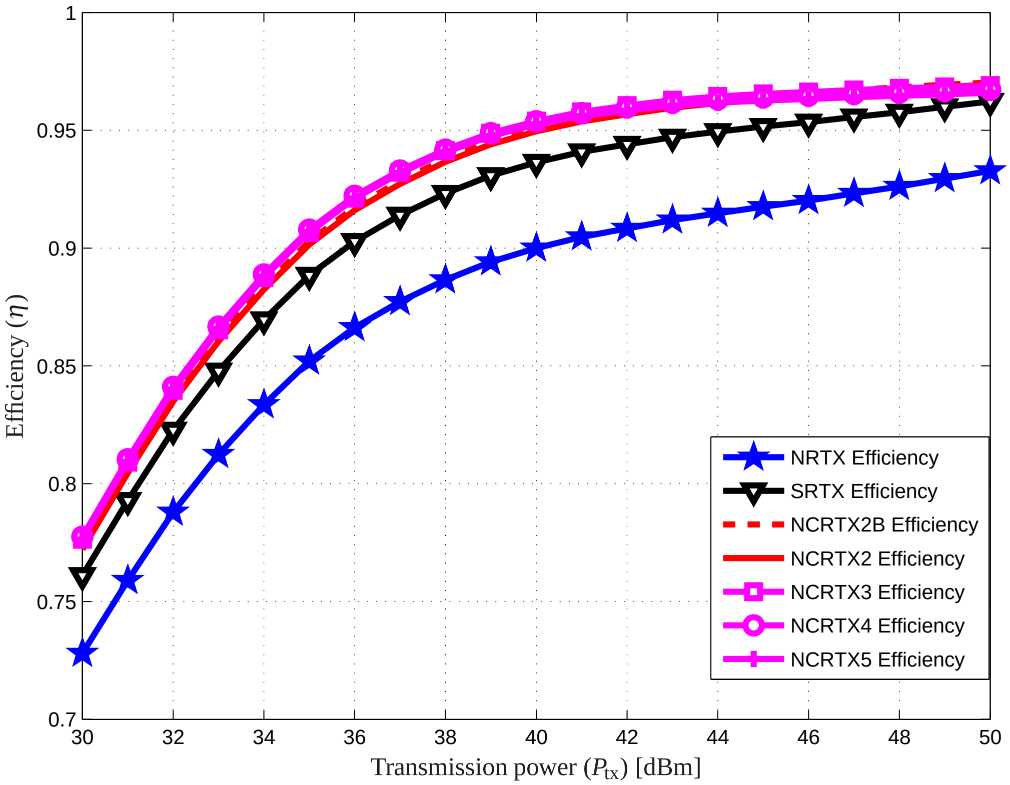


Figure 4.5: Impact of the transmission power (P_{tx}) on the efficiency (η).

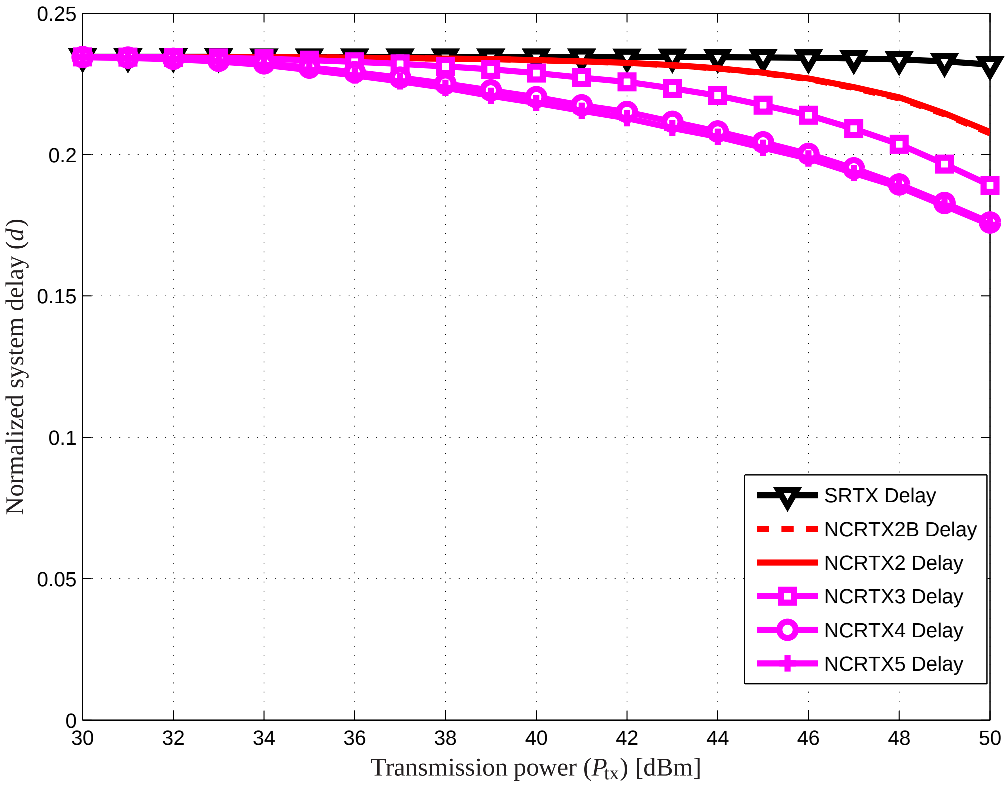


Figure 4.6: Impact of the transmission power (P_{tx}) on the normalized system delay (d).

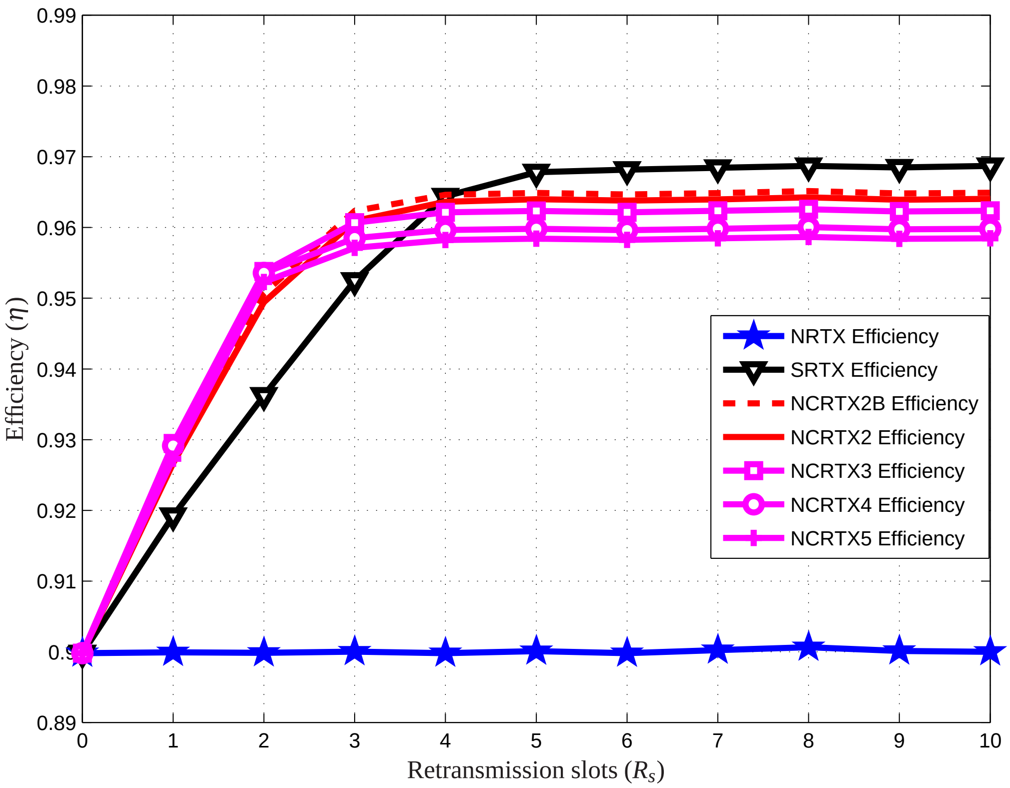


Figure 4.7: Impact of the number of reserved retransmission slots (R_s) on the efficiency (η).

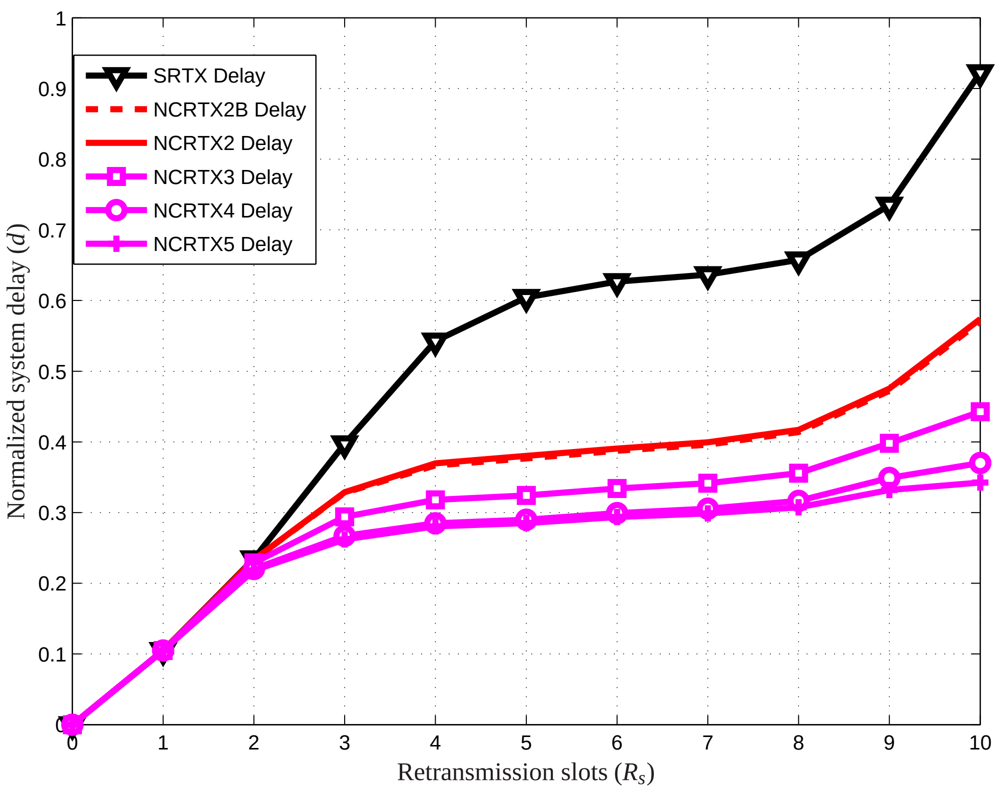


Figure 4.8: Impact of the number of reserved retransmission slots (R_s) on the normalized system delay (d).

enough to accommodate all retransmissions, thus not requiring the combining of packets in order to make them fit into the available number of retransmission slots.

The impact of the number of slots per frame, with R_s fixed, is shown in Figs. 4.9 and 4.10. The x-axis starts from 1 slot, but only in this case, the maximum number of retransmission slots per frame is limited by the size of the frame. For a large number of slots all algorithms approach the case without retransmission, since R_s is fixed. The achieved results are straightforward and confirm the importance of properly reserving at least a number of slots dedicated for retransmission.

Finally, the last results are shown in Figs. 4.11 and 4.12, representing the efficiency and normalized system delay as a function of the tolerance. The x-axis ends at 35 %, because around this value the performance of all algorithms converge to the case without retransmission. As of 10 % of tolerance, all retransmission algorithms have a performance very similar with respect to the metric of efficiency. But even though all retransmission algorithms have the same delay for the case without tolerance, in its presence the better performance of the algorithms with multiple coded packets is fairly visible.

4.2 Impact of imperfect self-interference cancellation for bi-directional relaying

This section describes the simulation model of the impact of imperfect self-interference cancellation for bi-directional relaying and a performance analysis and its numerical results are shown.

4.2.1 Simulation model

In the simulation model, all RSs have a different MS group that depends on the path-loss between RSs and MSs. Each MS chooses the RS which has the lowest path-loss to join the MS group of this RS.

The path-loss is mainly a function of distance between the MS and the RS or the RS and the BS. It was modeled according to [27] as follows:

$$PL = 114 + 35\log(2d)[dB], \tag{4.3}$$

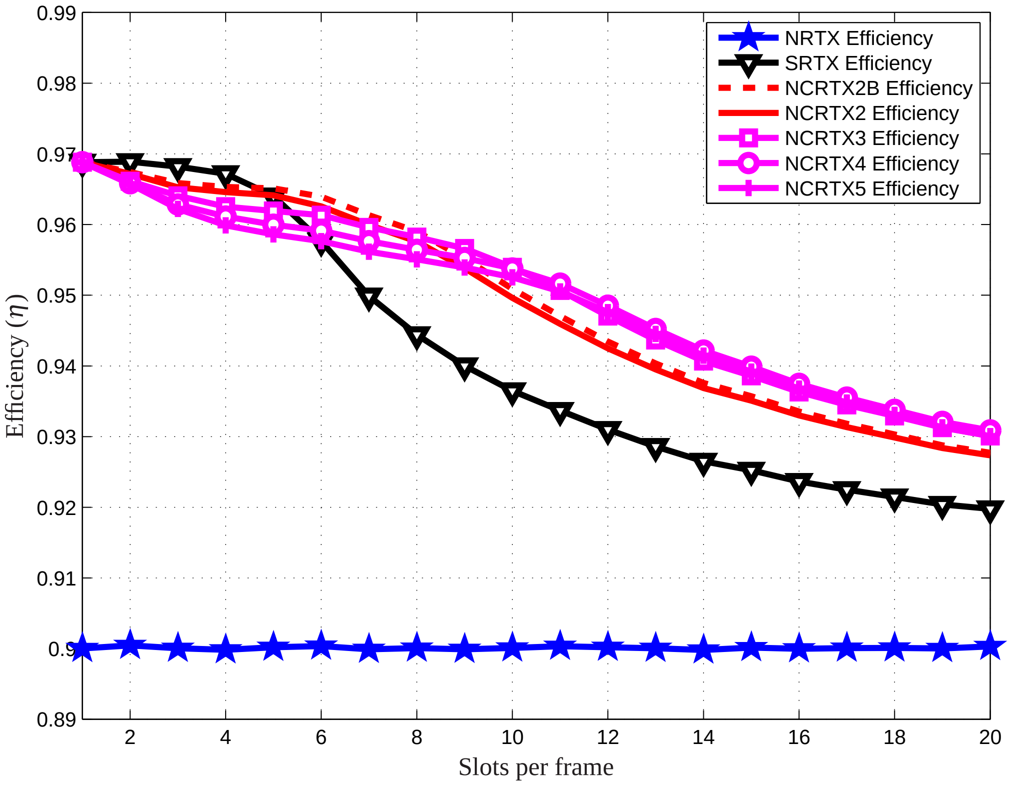


Figure 4.9: Impact of the number of slots per frame on the efficiency (η).

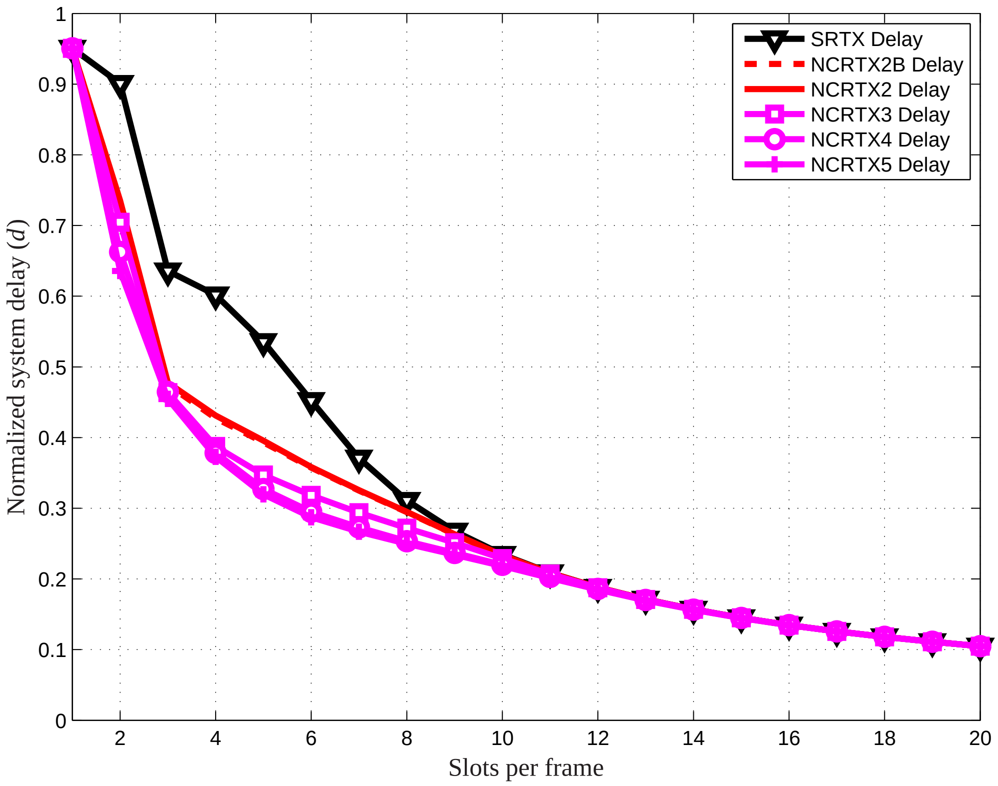


Figure 4.10: Impact of the number of slots per frame on the normalized system delay (d).

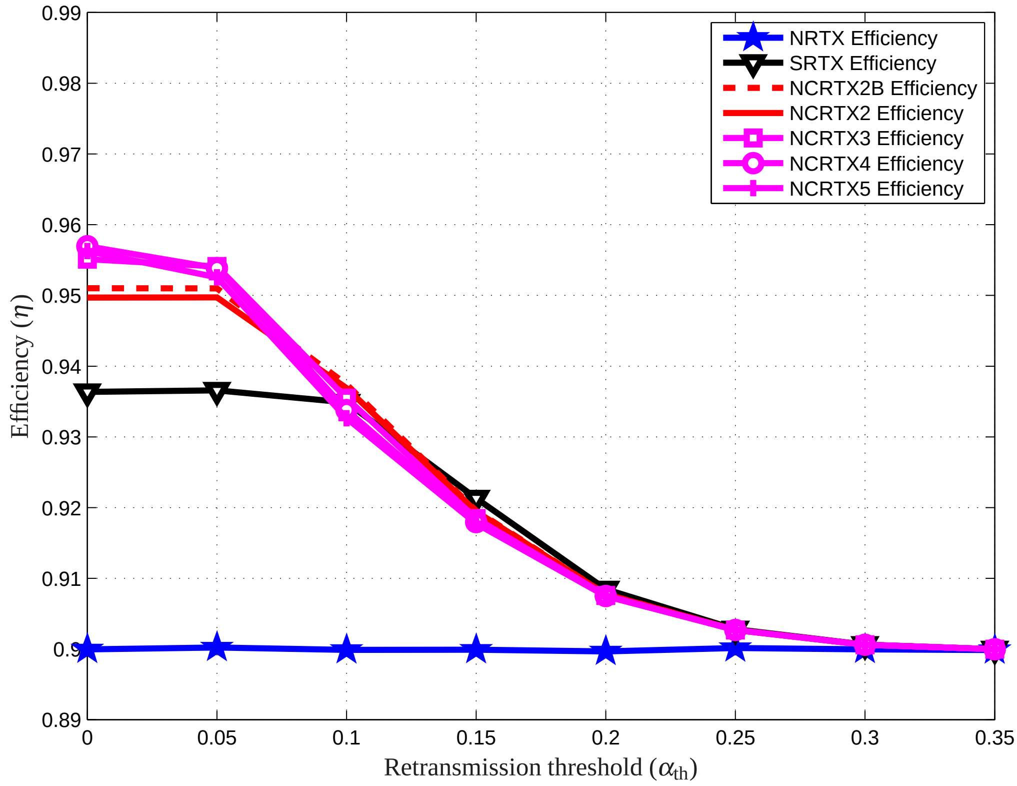


Figure 4.11: Impact of the retransmission threshold (α_{th}) on the efficiency (η).

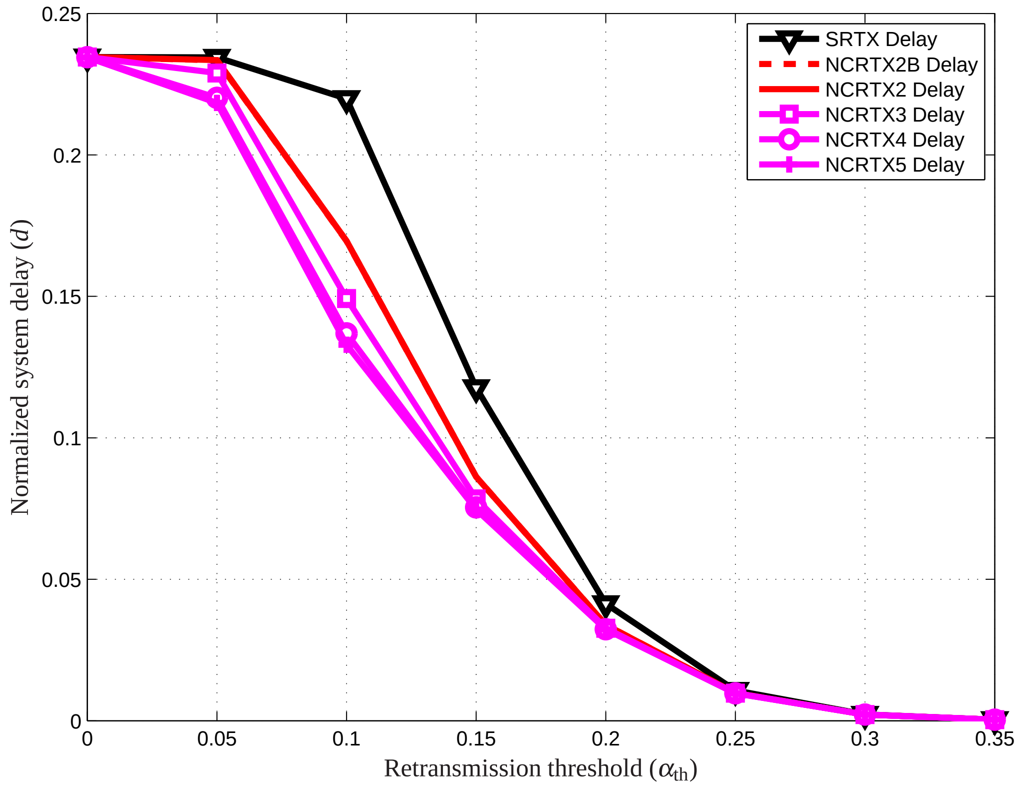


Figure 4.12: Impact of the retransmission threshold (α_{th}) on the normalized system delay (d).

Table 4.2: *List of considered system simulation parameters for the relaying scenario.*

Parameter	Value	Unit
Number of MSs (N)	10	-
Number of RSs	3	-
Cell radius	0.5	km
RSs ring radius	0.25	km
Transmission power of MSs	24	dBm
Transmission power of RSs	34	dBm
Transmission power of BS	34	dBm
Shadowing standard deviation	8	-
Noise power	-100	dBm
Iterations	10,000	-

where PL is the path-loss and d the distance.

Afterwards, both shadowing and fast fading are added to PL, according to 3.3, to calculate SNRs corresponding to the equivalent transmissions from BS to MSs and from MSs to BS. The shadowing or slow fading can be caused when a large obstruction such as a hill or large building obscures the main signal path between the MS and the RS or the RS and the BS. The amplitude change caused by shadowing was modeled using a log-normal distribution.

4.2.2 Numerical results and analysis

A simulation tool was built to evaluate several different scenarios and algorithms concerning the provision of AF bi-directional communication relaying services. The standard simulation parameters can be found in Table 4.2. In order to quantify the performance of the considered algorithms, the following key parameters have been chosen for analyzing their behavior maintaining the values of the other parameters according to Table 4.2:

i. Number of MSs within the cell (N);

ii. Total mumber of RSs;

iii. Cell radius;

iv. RSs ring radius.

The first parameter was chosen in order to analyze the impact of an increased load on the AF bi-directional communication relaying system. The variation of both second and third parameter has an impact on the radio link quality and refers to the convergence of the simulation results. Finally, the variation of the last parameter provides some insights on how to properly adjust the positioning of the RSs.

Note that the inferior bounding curve, which has no estimation and is shown for comparison purposes, corresponds to a worst-case in terms of C regardless of the variation of all key parameters. The superior bounding curve, which has perfect estimation and also is shown for comparison purposes, always has a better performance, or at least the same, for the case of large cells.

Fig. 4.13 represents the graphic of the sum rate (C) as a function of the number of MSs (N). For all algorithms, in spite of the increasing system load, the sum rate (C) remains constant.

The inferior bounding curve which considers $\hat{\beta}_r = 0$ has the sum rate (C) roughly 2 bit/s/Hz. The first algorithm which takes into account only the path-loss to calculate the estimated channel in $\hat{\beta}_r$ has the sum rate (C) roughly 3 bit/s/Hz. The second algorithm which corresponds to different ways of estimating $\hat{\beta}_r$ has the sum rate (C) roughly 4 bit/s/Hz.

Whereas, the third and last algorithm which both path-loss and shadowing are used to calculate the estimated channel in $\hat{\beta}_r$ has the sum rate (C) roughly 5 bit/s/Hz and the superior bounding curve which considers $\hat{\beta}_r = \beta_r$ has the sum rate (C) roughly 6.5 bit/s/Hz.

Since the number of required time slots is fixed in a total of $2N$, the increase in the number of MSs does not end up overloading the system, i.e., there are enough resources to transmit all data. The achieved results are straightforward and confirm the importance of properly estimating $\hat{\beta}_r$.

The impact of the number of RSs on the sum rate (C) of the algorithms is shown in Fig. 4.14. When there is only one RS, the C corresponds to a worst-case for all algorithms, since no grouping of MSs takes place, while for larger numbers of RSs C increases.

For large numbers of RSs the impact of this no grouping of MSs tends to be minimized, i.e., there is a diversity improvement of RSs and therefore a convergence to average values is perceived for each algorithm and metric. The relative performance among the algorithms is the same as that of Fig. 4.13 for the case with 10 MSs, due to the previously explained reasons. Note that as convergence is achieved, the advantage of algorithm 3 over the others becomes more evident.

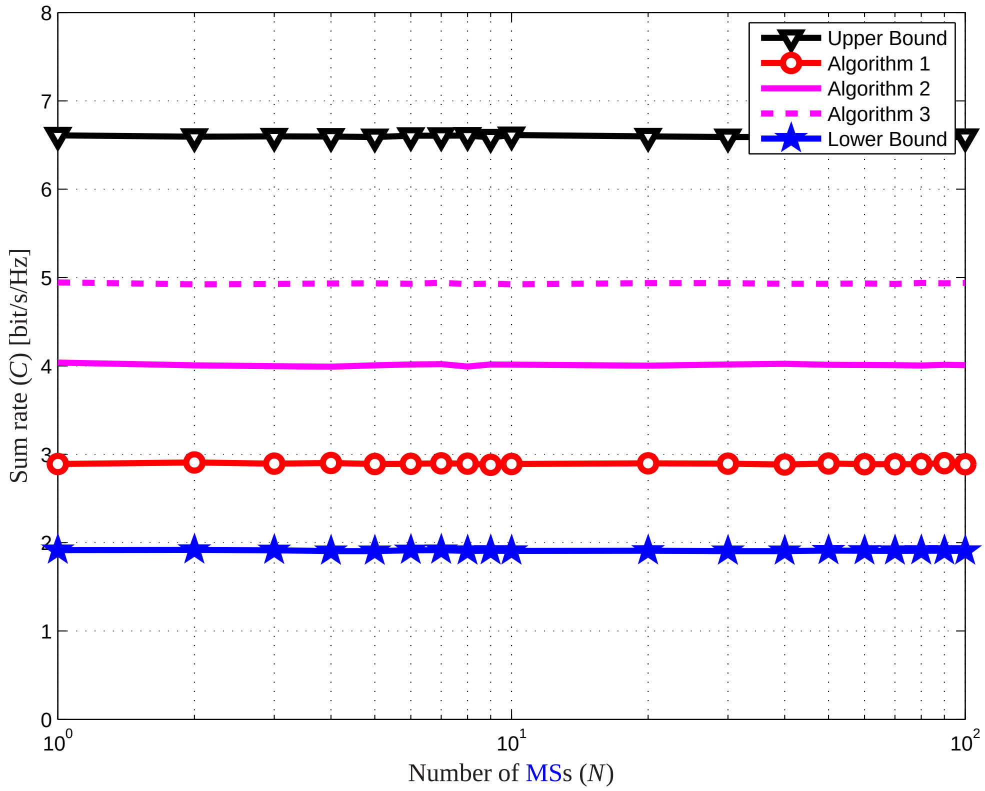


Figure 4.13: Impact of the number of MSs (N) on the sum rate (C).

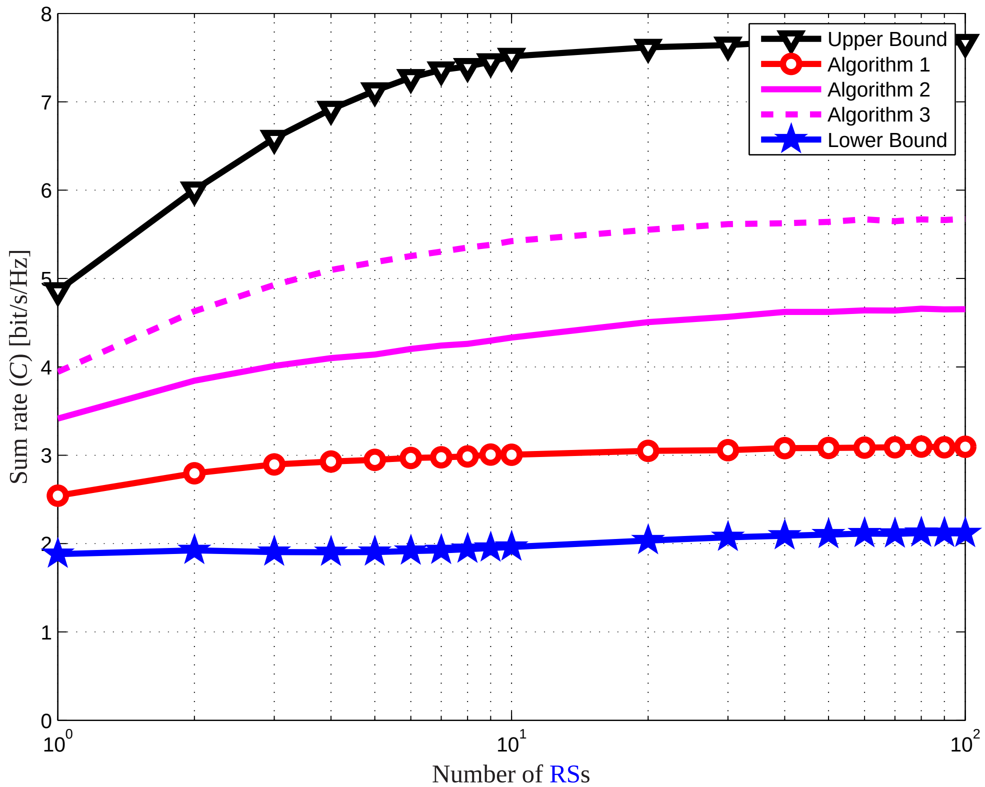


Figure 4.14: Impact of the number of RSs on the sum rate (C).

Fig. 4.15 shows the impact of the cell radius. Since the RSs ring radius is constant at 0.25 km, the x-axis starts from 0.3 km and, in this case, the minimum cell radius is limited by the size of the circumference of RSs radius, i.e., RSs are always within the cell.

Moreover, increasing the cell radius is never beneficial for the SNRs between the MSs and the BS because there is a uniform spatial distribution of MSs in every cell and, as the size of the RSs ring radius is fixed, the distance between the MSs and the RSs always increases. As for the achieved results of the sum rate (C) of the algorithms, they converge to small value for large cells sizes. This means that this cell radius is not already enough to accommodate all transmissions.

Fig. 4.16 illustrates the sum rate (C) as a function of the RSs ring radius. Note that the RSs ring radius has been simulated up to the cell radius. So, there is not possible to have the distance between RSs and the BS larger than the distance between MSs and the BS, i.e., in this case the RSs would be outside the cell.

It can be seen that the C of the superior bounding curve increases up to the RSs ring radius equal to 0.1 km while other algorithms decreases.

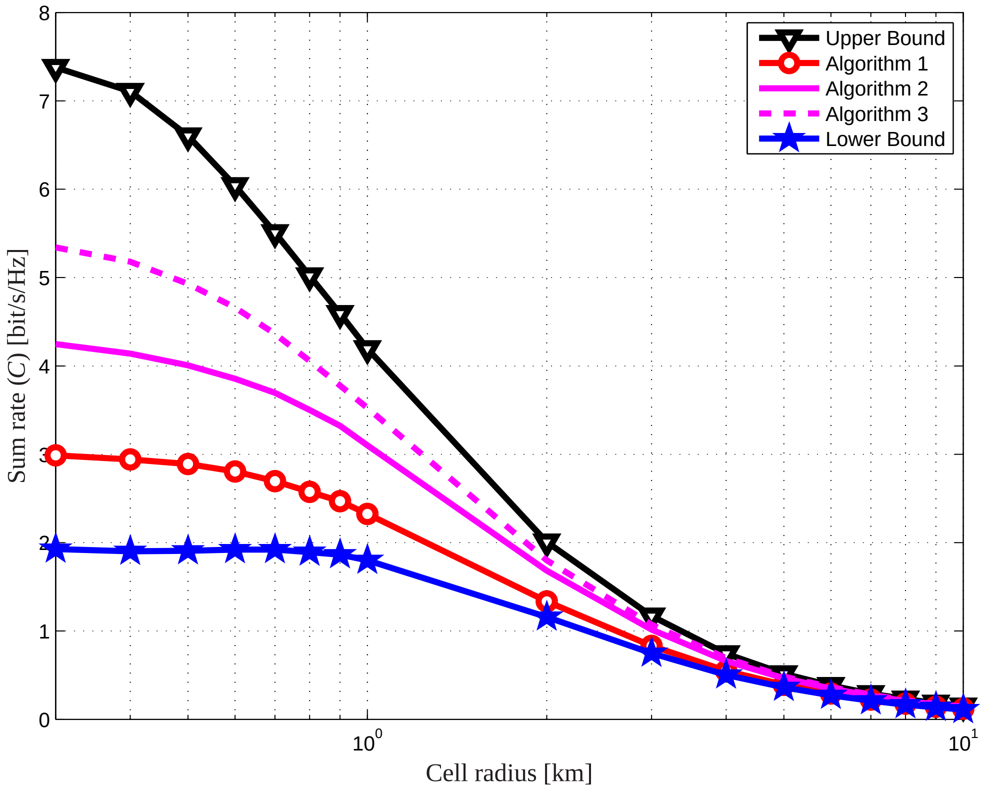


Figure 4.15: Impact of the cell radius on the sum rate (C).

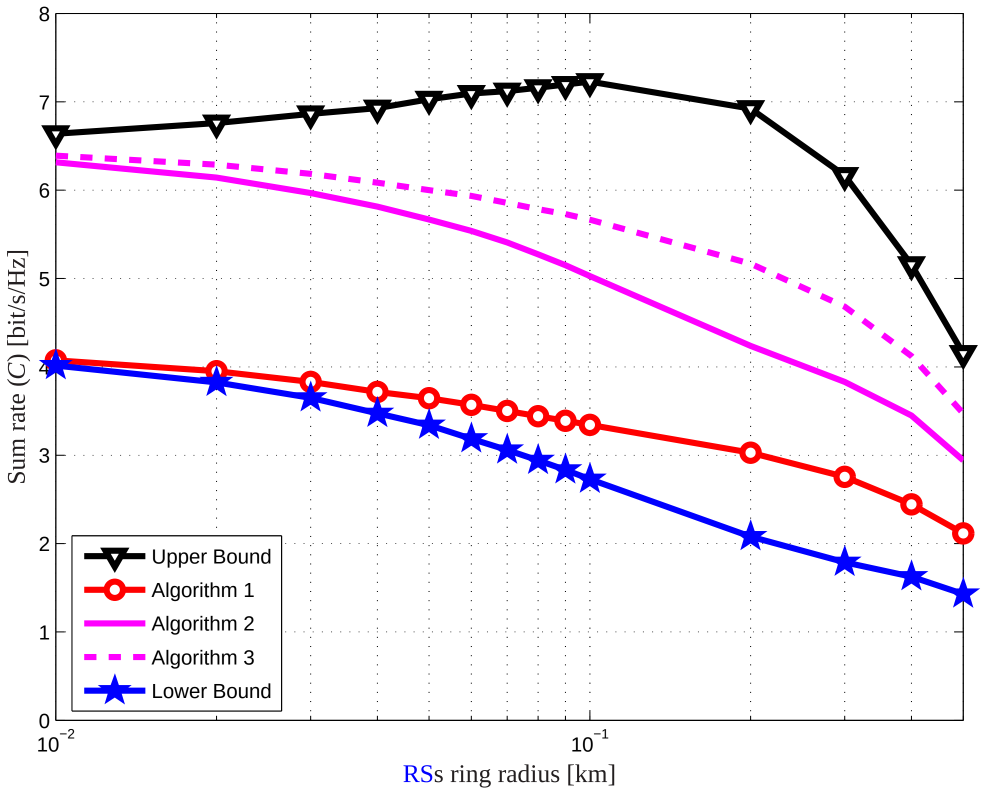


Figure 4.16: Impact of the RSs ring radius on the sum rate (C).

Chapter 5

Conclusion

5.1 Multicast retransmission techniques based on multiple coded packets

In this work, different types of multicast transmission schemes are analyzed and an algorithm based on network coding with a buffer is proposed. The proposed algorithm aims at using more efficiently the radio resources available for retransmission and it is shown to provide the best performance and significantly improve the efficiency in almost all cases, with the exception of rather small multicast group sizes. The impact of several system parameters has also been analyzed, in order to provide a detailed comparison among the algorithms.

Fig. 5.1 summarizes the gains provided by the proposed algorithm, taking into account the standard simulation parameters in Table 4.2. Gains of roughly 1.5% with regard to the case with simple retransmission and 5% with regard to the no retransmission algorithm are achieved. Note that higher gains are expected for larger multicast group sizes as well as for scenarios with less robust transmission schemes, i.e., with more pessimistic SINR to PEP mapping.

An interesting continuation of the current work is to extend the analysis to multi-cell scenarios. RSs implementing network coding, as well as combination of network coding and incremental redundancy for multicast services, are other subjects that might profit from efficient retransmission algorithms.

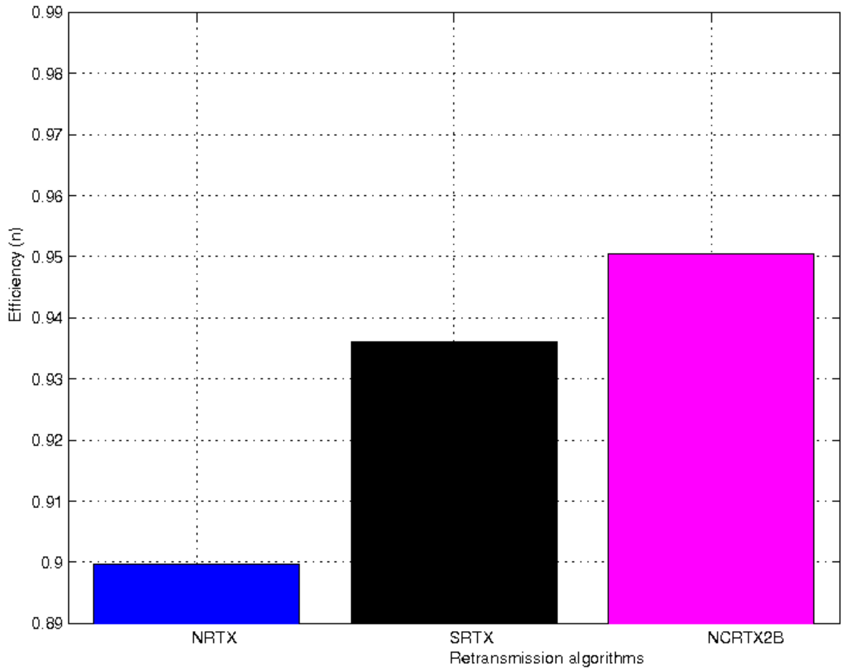


Figure 5.1: Comparison between the algorithms in terms of efficiency (η).

5.2 Impact of imperfect self-interference cancellation for bi-directional relaying

In this work, three types of error estimation through an imperfect SI cancellation for AF bi-directional communication relaying schemes are analyzed and these impact is evaluated. An efficient transmission technique for bi-directional communication services based on RS is compared to other schemes.

All algorithms aim at using more efficiently the available radio resources, i.e., the better the error estimation, of course, the better the performance. For rather large cells sizes the performance decreases very, making no difference the channel estimation. The impact of several system parameters has also been analyzed, in order to provide a detailed comparison among the algorithms.

The analysis showed that the way the channel is estimated has a significant impact on performance. The strategy of algorithm 3 has the best cost-benefit in terms of performance vs. complexity estimation.

A possible prospect for future works is to analyze the bi-directional communication considering DF relaying, instead of AF, as well as extend the analysis to multi-cell scenarios

Bibliography

[1] 3GPP, "LTE physical layer - general description," Third Generation Partnership Project, Tech. Rep. TR 36.201 V8.3.0, Mar. 2009.

[2] ——, "Applications of network coding in LTE-A," Technical Specification Group Radio Access Networks, Tech. Rep. R1 084130, Nov. 2008.

[3] ——, "Optional retransmission schemes for LTE-A MBMS," Technical Specification Group Radio Access Networks, Tech. Rep. R1 084132, Nov. 2008.

[4] M. Bakhuizen and U. Horn, "Mobile broadcast/multicast in mobile networks," *Ericsson Review*, issue no. 01/2005, May 2005.

[5] 3GPP, "Multimedia Broadcast/Multicast service (MBMS); architecture and functional description," Technical Specification Group Services and System Aspects, Tech. Rep. TS 23.246 v6.10.0, Jun. 2006.

[6] M. Rossi, M. Zorzi, and F. H. P. Fitzek, "Investigation of link layer algorithms and play-out buffer requirements for efficient multicast services in 3G cellular systems," in *Proc. IEEE PIMRC*, vol. 3, Sep. 2004, pp. 2256–2261.

[7] W. Chen, K. B. Letaief, and Z. Cao, "Opportunistic network coding for wireless networks," in *Proc. IEEE ICC*, Jun. 2007, pp. 4634–4639.

[8] H. Yomo and P. Popovski, "Opportunistic scheduling for wireless network coding," in *Proc. IEEE ICC*, Jun. 2007, pp. 5610–5615.

[9] S. Katti, H. Rahul, W. Hu, D. Katabi, M. Médard, and J. Crowcroft, “XORs in the air: Pratical wireless network coding,” *IEEE/ACM Trans. Networking*, vol. 16, no. 3, pp. 497–510, Jun. 2008.

[10] C. Fragouli, D. Katabi, A. Markopoulou, M. Médard, and H. Rahul, “Wireless network coding: Opportunities & challenges,” in *Proc. IEEE MILCOM*, Oct. 2007, pp. 1–8.

[11] D. Silva, F. R. Kschischang, and R. Koetter, “A rank-metric approach to error control in random network coding,” *IEEE Transactions on Information Theory*, vol. 54, no. 9, pp. 3951–3967, sep. 2008.

[12] D. Silva, F. R. Kschischang, and R. Kotter, “Capacity of random network coding under a probabilistic error model,” in *Proc. 24th Biennial Symposium on Communications*, Jun. 2008, pp. 9–12.

[13] D. Silva and F. R. Kschischang, “Using rank-metric codes for error correction in random network coding,” in *Proc. IEEE International Symposium on Information Theory (ISIT)*, Jun. 2007, pp. 796–800.

[14] S. Redana, K. Doppler, D. Schultz, M. Wódczak, P. Rost, L. Hamm, M. Döttling, and Q. Perez, “Final assessment of relaying concepts for all CGs scenarios under consideration of related WINNER L1 and L2 protocol functions,” WINNER, Tech. Rep. D3.5.3, Sep. 2007.

[15] T. Unger, C. Mueller, and A. Klein, “Resource allocation schemes for infrastructure based relay networks,” in *Proc. International OFDM Workshop (INOWO)*, Aug. 2006.

[16] C. Mueller, A. Klein, F. Wegner, and M. Kuipers, “Costs and performance of non-cooperative relay networks,” in *Proc. European Wireless Conference (EW)*, Apr. 2007.

[17] D. Nguyen, T. Tran, T. Nguyen, and B. Bose, “Wireless broadcast using network coding,” *IEEE Transactions on Vehicular Technology*, vol. 58, no. 2, pp. 914–925, Feb. 2009.

[18] P. Popovski and H. Yomo, “Bi-directional amplification of throughput in a wireless multi-hop network,” in *Proc. IEEE Vehicular Technology Conference (VTC)*, vol. 2, May 2006, pp. 588–593.

[19] T. Unger and A. Klein, “On the performance of two-way relaying with multiple-antenna relay stations,” in *Proc. IST Mobile and Wireless Communications Summit*, Jul. 2007.

[20] ——, “Duplex schemes in multiple antenna two-hop relaying,” *EURASIP Journal on Advances in Signal Processing*, vol. 2008, pp. 1–14, 2008.

[21] S. Berger, M. Kuhn, A. Wittneben, T. Unger, and A. Klein, “Recent advances in amplify-and-forward two-hop relaying,” *IEEE Communications Magazine*, pp. 50–56, Jul. 2009.

[22] A. Amah, A. Klein, Y. C. B. Silva, and A. Fernekess, “Multi-group multicast beamforming for multi-user two-way relaying,” in *Proc. ITG/IEEE Workshop on Smart Antennas (WSA)*, Feb. 2009, pp. 53–60.

[23] J. Manssour, A. Osseiran, and S. B. Slimane, “Wirelss network coding in multi-cell networks: analysis and performance,” in *Proc. IEEE International Conference on Signal Processing and Communication Systems (ICSPCS)*, Dec. 2008, pp. 1–6.

[24] A. Osseiran, K. Doppler, C. Ribeiro, M. Xiao, M. Skoglund, and J. Manssour, “Advances in device-to-device communications and network coding for IMT-advanced,” in *Proc. ICT Mobile Summit*, Jun. 2009, pp. 1–8.

[25] P. Larsson, N. Johansson, and K.-E. Sunell, “Coded bi-directional relaying,” in *Proc. IEEE Vehicular Technology Conference (VTC)*, vol. 2, May 2006, pp. 851–855.

[26] 3GPP, “Physical layer aspects for evolved universal terrestrial radio access (UTRA),” Third Generation Partnership Project, Tech. Rep. TR 25.814 V7.1.0, Sep. 2006.

[27] J. Manssour, A. Osseiran, and S. Ben Slimane, “Wireless network coding in multi-cell networks: Analysis and performance,” in *Proc. IEEE International Conference on Signal Processing and Communication Systems (ICSPCS)*, Dec. 2008, pp. 1–6.