University of Strathclyde

Department of Electronic and Electrical Engineering

Cooperative Network Coding for Wireless Networks

By

Hani Hasan Attar

A thesis presented in fulfilment of the requirements for the degree of Doctor of Philosophy

2011

Dedication

Dedicated to

My parents and my wife for their love and support

Declaration

This thesis is the result of the author's original research. It has been composed by the author and has not been previously submitted for examination which has lead to the award of a degree.

Hani Attar August 2011

Acknowledgements

It is always easy to know whom to thank after finishing such big work, but it is truly hard for me to express my feeling of being grateful toward them in few words, mainly because the words are much less than the favour I have received. However, I am sure that all of the people who helped me know how grateful I am, with good thanking words or even without naming them.

I would love to start thanking the people who made this thesis possible by the academic support they showed. So, I would like to thank my supervisor Dr Lina Stankovic for the support she showed through my research time, indeed, the endless ideas Dr Lina Stankovic provided me played the major role in this thesis. Moreover, I am grateful to her good understanding for my private situation. I will always be grateful for her.

Beside my supervisor, I would like to thank Dr Vladimir Stankovic for the help he showed mainly in proof reading my published papers. The fewer things I can remember in here is the good English language I learnt from Dr Vladimir Stankovic's papers beside the good knowledge in them.

To Dr Dejan Vukobratovic, I would like to show my gratefulness for the good academic cooperation he showed.

Lastly, and most importantly, I owe my loving thanks to my mother and my wife Mirs Carole Meyer-Attar as without their understanding, endless patience and encouragement it would have been impossible for me to finish this work. Thank you my mother for the kind prayers and the financial support, thank you Carole for the time you spent with our two kids when I was away and for your kind support.

Abstract

We propose applying cooperative Network Coding (NC) over wireless networks; starting from the physical layer, we propose two practical power- and bandwidth-efficient systems based on amplify-and-forward (AF) and decode-and-forward (DF) schemes to address the problem of information exchange via a relay. The key idea is to channel encode each source's message by using a high-performance non-binary turbo code based on Partial Unit Memory (PUM) codes to enhance the bit-error-rate performance, then reduce the consumed energy and increase spectrum efficiency by using NC to combine individual nodes' messages. Simulation results under Additive White Gaussian Noise confirm that the proposed scheme achieves significant bandwidth savings and fewer re-transmissions over the benchmark system which does not resort to NC. Moreover, we propose a cooperative strategy that is useful when insufficient combined messages are received at a node to recover the desired source messages, which has been applied over Wireless Sensor Network (WSN) as a proposed application. Further we propose applying cooperative NC over MAC layer by proposing simple packet-level cooperative transmission protocols that increase efficiency of multiple WSN nodes in delivery of data to a joint central point destination node through the use of NC.

We start with a simple design of an erasure channel, assuming that all users to users channel with the same erasure probability, regardless the distances, and the same for all users to the destination, and then, full analysis is performed.

Next, we are applying NC cooperative over the downlink communication to prove that NC cooperative technique is as useful in the uplink, so, the NC cooperative scheme has been applied over LTE-A scenario where the Pico relay receives the data from the source and forwards them to the Femto relays, whose then forward them to the users (downlink scenario). Simulation results showed good improvement obtained in term of

packet error probability and transmission data rate. Moreover, it showed how significantly we improved the Automatic Repeat Request needed when NC is used.

For more practical research, Finite State Markov Chain has been implemented as a time varying Rayleigh block-fading channel in both uplink and down link scenarios.

Finally, we extended applying FSMC over Video transmission to obtain the results which can be used for future research when NC could be needed for further future work.

Publications

Author's publications in reverse chronological order.

Journal Publication

[J1] Sajid Nazir, Vladimir Stankovic, Hani Attar and Samuel Cheng 'Relay-assisted Rateless Layered Multiple Description Video Delivery' IEEE Journal on Selected Areas in Communications – Special Issue on Theories and Methods for Advanced Wireless Relays 2011. Submitted.

[J2] H Attar, L Stankovic, and V. Stankovic 'A cooperative network coding system for wireless sensor Networks'. Final Acceptance has been confirmed in IET Proceedings in Communications 2011. This work is described in Chapter 3.

Conference Publications

[C4] H. Attar, D. Vukobratovic, L. Stankovic and V. Stankovic, "Performance analysis of node cooperation with network coding in wireless sensor networks," Proc. 4th IEEE Int. Conference on New Technologies, Mobility and Security (NTMS 2011), Feb. 2011, Paris, France.

[C3] H. Attar, L. Stankovic, V. Stankovic and C. Khirallah, "Physical Layer Multi-Source Packet Network Coding with PUM Turbo Codes," 3rd IEEE Mosharaka Intl Conf on Communications, Computers and Applications (MIC-CCA-2009), October 2009, Amman, Jordan

[C2] H. Attar, C. Khirallah, L. Stankovic and V. Stankovic, “Physical layer multi-source network coding using PUM turbo codes,” Proc. 10th International Symposium on communication Theory and Applications (ISCTA 09), Ambleside, Lake District, U.K, July 2009.

[C1] H. Attar, L. Stankovic and V. Stankovic, “Physical-layer network coding based on PUM turbo codes,” Proc. IEEE Mosharaka International Conference on Communications, Signals and Coding, Amman, Jordan, February 2009.

Papers 2, 3 and 4 are described in Chapter 3. Paper 4 is partly described in Chapters 4 and 5. Paper 1 is partly described in Chapter 5.

Contents

List of Figures

List of Tables

Abbreviations

ACK	Acknowledgement
AF	Amplify-and Forward
AF_b	Benchmark Amplitude-and-Forward system
AF_p	Proposed Amplitude and Forward
ARQ	Automatic Repeat Request
AL-FEC	Application-Layer Forward Error Correction
AWGN	Additive White Gaussian Noise
BPSK	Binary Phase Shift Keying
BS	Base Station
CoMP	Coordinated Multi-Point
CSMA-CA	Carrier Sense Multiple Access with Collision Avoidance
DF	Decode-and-Forward
DF_b	Benchmark Decode-and-Forward
DF_p	Proposed Decode-and-Forward
DL	Down Link
DVB-NGH	Digital Video Broadcast- Next Generation Handheld
DSL	Digital Subscriber Line
ECC	Error Correcting Code
FEC	Forward error correction
FSMC	Finite-State Markov Chain

GEA	Gaussian Elimination Algorithm
GJEA	Gauss-Jordan Elimination Algorithm
GTS	Guaranteed Time Slot
HARQ	Hybrid Automatic Repeat Request
HES	High Error Set
IMT-A	International Mobile Telecommunications Advanced
ITUA	Intrusion Tolerance by Unpredictable Adaptation
IEEE	Institute of Electrical and Electronics Engineers
ISM	Industrial Scientific Medical
LDPC	Low Density Parity Check
LES	Low Error Set
LQI	Link Quality Indication
LTE	Long term evaluation
LTE-A	Long term evaluation advanced
MAC	Media Access Control
MANET	Mobile Ad-hoc Network
MDC	Multiple Description Coding
NPSD	Noise Power Spectral Density
NAK	Non-Acknowledgements
NC	Network Coding
O-QPSK	Offset-Quadrate Phase Shift Keying
OSI	Open Systems Interconnection
PDF	Probability Density Function
PEP	Packet Error Probability

PER	Packet Error Rate
PLNC	Physical Layer Network Coding
PHY	Physical
PUM	Partial Unit Memory
PUMTC	Partial Unit Memory Turbo Code
P2P	Peer-to-Peer
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
RLC	Random Linear Codes
RSSI	Received Signal Strength Indicator
SNR	Signal to Noise Ratio
RSC	Systematic Convolutional Code
TDMA	Time-Division Multiple Access
TC	Turbo Code

List of Symbols

a and b	Uncoded data
X_{En}	Coded data
N	Number of nodes or users in the network
a_{rec} and b_{rec}	Reconstructed transmitted data
$X_{\mathrm{ret\text{-}En1}}$	Encoded reconstructed data
Y_{rec}	Received signal
$\mathrm{A}_{\mathrm{DFb}}$	Amplitude Y_{BS} for DF_{b}
Z_{i}	Additive white Gaussian noise for channel i
Y_{BS}	Received signal at the base station
C	Channel capacity
R	Transmission rate
K	Code constraint length
d_{fre}	Minimum weight of a codeword
M_{r}	Number of memory registers
μ	Number of bits in the shift register
C_t	Output word of PUM code
u_t	Current information bits
E_b/N_0	Bit-to-noise
σ^2	Variance
$L_e(u)$ ratio	Extrinsic Log-likelihood

P	Partitioning Matrix
I	Identity Matrix
G(D)	Generator Matrix with delay D
L(u)	Log-likelihood ratio
f_c	Carrier frequency
v	Speed
C	Speed of light = $3 \cdot 10^8$ ms^{-1}
$P_r(d)$	Received power
G_r	Received antenna Gain
G_t	Transmitted antenna gain
λ	Wavelength
f_D	Doppler frequency
z^{UL}	Uplink Gaussian noise
z^{DL}	Downlink Gaussian noise
X_c	Transmitted combined packet
$\hat{Y}c$	Received combined signal
M	Number of users in cooperative system
D	Destination
q	Erasure probability
C_r	Received combined packet
P_S	Probability for selfish mode
p_c	Probability for cooperative mode
r	Transmission distance

$P_{d,2}$	Conditional probability for successful decoding after the second stage
PER	Packet Error Rate
$C_{2-odd-even}$	Two packets combined in odd even order
P_e	Probability for un-successful decoding
T_P	One packet time period
f_m	Maximum Doppler frequency
$P_{k,i}$	Packet transition probability between states k and i
C_k	Constant criterion for FSMC
Γ_k	Γ_i SNR partition level for state-space
γ_0	Average SNR
π_k	Steady state probabilities π_i
$N(\Gamma_k)$	Level crossing rate for stage k
L	Number of bits per packet
P_{ek}	Symbol error rate
f_k	BER for state k
$R_{DR(k)}$	Received data rate for state k

1. Introduction

Wireless network refers to any type of network that is not connected by cables of any kind i.e., nodes are connected by broadcast propagation over radio waves transmission system.

Wireless Personal Area Networks (WPANs), Wireless Local Area Network (WLAN), Wireless Metropolitan Area Networks (WMAN), Wireless Wide Area Networks (WAN), and Mobile devices networks are all types of wireless networks.

Modern technology is shaped by the need for wireless communication devices (e.g. wireless sensors, cell phones, laptop, etc). Moreover, the mobility demand for these communication devices makes our modern technology more and more related to wireless network communication. The high growing pace for wireless communication is justified by the mobility, the ease of use, the flexibility of installation, and the reasonable cost. All these features make movable wireless network technology nowadays to be applied almost in everywhere, such as trains, busses and even in personal laptops. In fact, we can claim that mobile network covers everywhere in the UK for example. The traditional wired networks are limited in term of fulfilling these demands. This is beside of the implementation cost for wired networks, which enables wireless network to be a better option even in term of cost and implementation.

This demand on wireless communication has been directing a considerable number of researches towards wireless networking technologies such as Wireless Sensor Network (WSN), [1], Mobile Ad-hoc Networks (MANETs) [2], cellular networks [3], WLANs [4], Ultra-WideBand (UWB) networks [5], Institute of Electrical and Electronics Engineers (IEEE) standards for WLAN IEEE 802.11, WiFi IEEE 802.11a , PAN IEEE 802.15, WiMax IEEE 802.16e-2005, Third Generation Partnership Project (3GPP), long Term Evaluation (LTE), and LTE-Advanced (LTE-A). Among these, WSN, PAN, LTE-A and mobile ad-hoc networks are widely used and being researched. The wireless communication applications and design are mainly related to the power consumption and communication range, i. e., WSN is power limited network; therefore, it is only targeted at low power short-range wire replacement, unlike 3GPP, LTE, and LTE-A that target applications with better power sources and much longer ranges, and supported by different types of base stations and relays.

Further improvement in wireless networking is needed in WSN to allow the network's nodes to participate in relaying the traffic for other nodes, i.e., enabling wireless multi-hop communications.

In fact relaying the traffic goes even beyond the benefit of power consumption; it is also used for long range communications such as LTE-A to improve the channel capacity and Packet Error Rate (PER) in lossy channels, which enhances the communication's quality and reliability. Modern techniques nowadays play important roles to improve the communication range, data rate, bandwidth, and the power efficiency such as Network Coding (NC) [6] which shows how NC results to bandwidth saving and increase in the throughput, and Cooperation Networks [7] where two or more users in a network can share their information and jointly transmit their messages to obtain better reliability and efficiency than they could have when transmitting their information separately. Beside these techniques, Forward Error Correction (FEC) has obtained significant attention in recent years.

Network coding is a new technique which works with many other techniques and can be applied over different layers for different applications. So, to fully understand our contributions, it is important to introduce these techniques and show how they are related to our research work.

We are applying NC and cooperation over wireless network to improve the reliability and to save the bandwidth resulting to increase the throughput; this makes it important to introduce both NC and cooperative techniques in this thesis. Moreover, as our contributions are directed to the wireless network broadcasting, it is important as well to understand the nature of broadcasting and the free space broadcasting together with the free space losses and channels, such as Additive White Gaussian Noise channel (AWGN) and fading channel.

In addition, we have applied our contribution over different applications, such as WSN and LTE-A mobile network, which justifies the reason we introduced the background for these applications including the wireless topologies and LTE-A aspects such as Hybrid Automatic Repeat Request (HARQ) and Coordinated Multi-Point (CoMP).

To make our erasure channel more practical in term of time varying channel, Finite-State Markov Chain (FSMC) has been implemented, and as a result, our back ground has been extended to cover this vital technique in good details showing the way it has been simulated and the full details for the theoretical bounds, such as concepts and theoretical equations in the addition to the full details of the FSMC to represent the Rayleigh block-fading channel in the wireless communications.

Finally, we have used Partial Unit Turbo Code (PUTC) as our FEC code over the physical layer, so, we introduced this code in good details to show how it performs the error correction and what error correction performance it provides when applying NC over the transmitted data.

1.1 Network Coding

Network Coding [6] is a novel technique originally proposed for multicasting information over wireline networks of noiseless channels. It is based on combining received information packets; that is, each relay or Base Station (BS) node computes a certain encoding function of the received packets and forwards the resulting packet towards its destination.

Figure 1.1 shows the traditional method without applying NC and with applying NC. Figure 1.1 illustrates, when applying NC in the BS, the number of necessary downlink transmissions from the intermediate node (BS) to the two users, is reduced from two to one, and thus the throughput is increased by 50%.

Suppose that two users 1 and 2 want to exchange their information packets via a BS (or relay node). Both users 1 and 2 generate their messages *a* and *b* respectively, and then encode them to X_{En1} and X_{En2} before sending over an uplink wireless channel to the BS. Then the role of the BS is to relay the signals it has received, which contrasts with traditional communications via two orthogonal downlink channels.

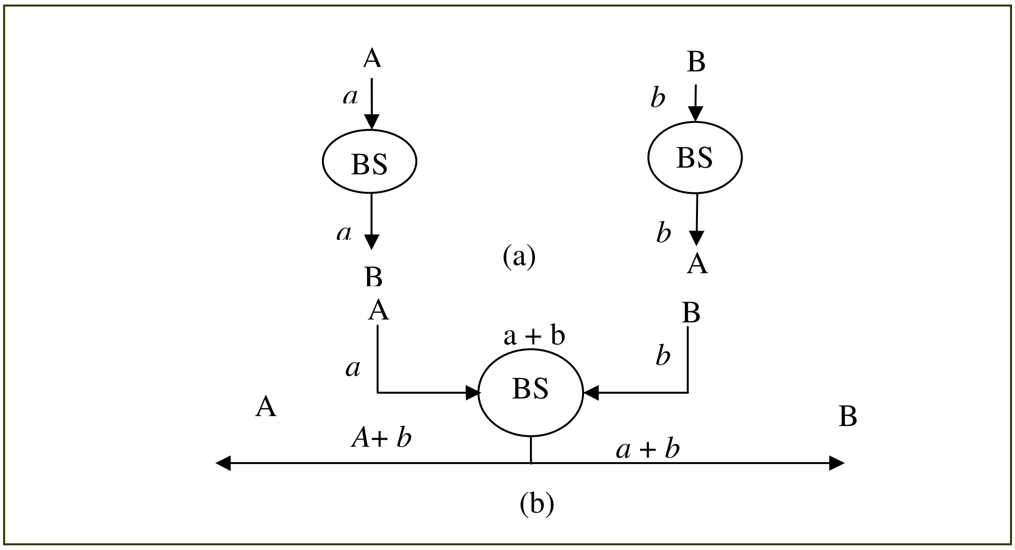


Figure 1.1: (a) Base station without applying Network Coding, (b) Base station with applying Network Coding.

So, instead of transmitting to X_{En1} and X_{En2} separately, the BS can broadcast (X_{En1}+ X_{En2}) over the wireless radio link, where the addition is a modulo-2 operation if the BS performs the decoding over the received data streams. Then User 1 subtracts b from the received and then decoded (possibly corrupted) $X_{\text{En1}} \oplus X_{\text{En2}}$ and reconstructs the desired message X_{ret1} where X_{ret1} is the retrieved message of a. Similarly, User 2 subtracts a and recovers X_{ret2}. Thus, only one transmission is needed, which decreases power consumption. As the bit stream length of ($X_{\text{En1}} \oplus X_{\text{En2}}$) packet is equal to the length of either X_{En1} or X_{En2}, the required transmitter power is half that is needed of two transmissions individual [8].

1.2 Network Coding Benefits

The benefits NC provides in communications have such a wide range; however, we summarize the most important ones that have been exploited in our research as follow:

- **Throughput Gain**: In multicast network, NC can improve the capacity, as nodes can use the information sources efficiently, resulting to maximum rate flow at the destination nodes [9] and [10].
- **Reliability:** Network Coding is used in order to maintain the network reliability, where every lost packet needs to be retransmitted. When applying NC, every

intermediate node can transmit multiple packets due to the linear combination. If a node does not successfully receive a packet combination, the original packet can still be decoded via getting the missing information through other linear combinations [10].

- **Mobility and Adaptability:** Network coding does not require knowledge of the network nor routing updates; this enables NC to be a very beneficial dynamical network. In such case, the source will perform the linear combinations of the packets and transmit them to the destination node. Destination then evaluates the received combined packets to confirm whether this combined packet has any novel piece of information or not.

Based on above, NC can be an efficient solution when compared with routing, where continuous routing updates are required [10]. However, the synchronization is considered as a difficult challenge at a high mobility speed. The reasons behinds the synchronization difficulties are related to the BS, as it is difficult for the BS to know what each combined packet is contained from.

1.3 Network Coding Applications in Wireless Network

The basic potential applications of NC can be summarised in the following:

- **P2P File Distribution and P2P Streaming:** In Peer-to-Peer (P2P) file distribution networks, NC would help to minimize the required downloading time in both live and on-demand P2P streaming. Additionally, in the case where a server stops transmitting before all peers fully download all files, the diversity of the blocks provides a more robust solution and it is the same for the case when peer nodes leave when their file has been downloaded. In a mesh network topology, NC's adaptability could improve the performance [9] and [11].
- **Wireless Networks:** The benefit of improving the throughput is one of the most important benefits of NC which is applied in wireless networks, mainly when the wireless nodes communicate via a common node (BS), such as mulit-hope wireless network. Furthermore, [9] shows that NC can improve the performance in wireless mesh network when experimented with 802.11 hardware, resulting to almost

doubling the throughput. Though the system that was under experiment is a simple XOR-ing combination of the packets, it is important to show it because we build on our work over it. Moreover, NC can be applied over the cooperative network when the broadcast packets could be overheard by the neighbouring nodes [7].

- **Ad-hoc Sensor Networks:** NC can be applied in Sensor Network in the case of untuned radios, to maintain the low complexity and price, as applying NC could replace the quatz oscillator in order to tune the radio to a specific frequency, this leads to avoid the higher cost and complexity of the hardware when quatz oscillator is used, i., e., instead of a quatz oscillator, an on-chip resonator combined with random NC is used and gives satisfactory performance [9].
- **Cellular Networks**

  Network Coding is being used in modern cellular networks, such as 4th generation mobile networks, as it shows significant improvement in data rate and diversity [12] and [13].

In our research, we applied NC over Ad-hoc sensor nodes for data gathering. Each sensor node receives a linear combination of packet and then provides it to the sink node. As a result, the sink nodes of the sensor network will benefit from the improvement of throughput [9].

1.4 Cooperation in Relay Networks

As a result of the broadcast nature of wireless communication, a source signal transmitted towards the destination can be ‘overheard’ at neighbouring nodes. Network Cooperation occurs when the neighbouring nodes process this overheard information and retransmit it towards the destination, resulting to higher throughput gained by the extra spatial diversity [7]. In the case of *N* users broadcasting data to a common destination, users can cooperate to improve the communication performance.

In such case, all users broadcast to the common destination in the first stage, and then wait for the confirmation that all packets have been received at the destination.

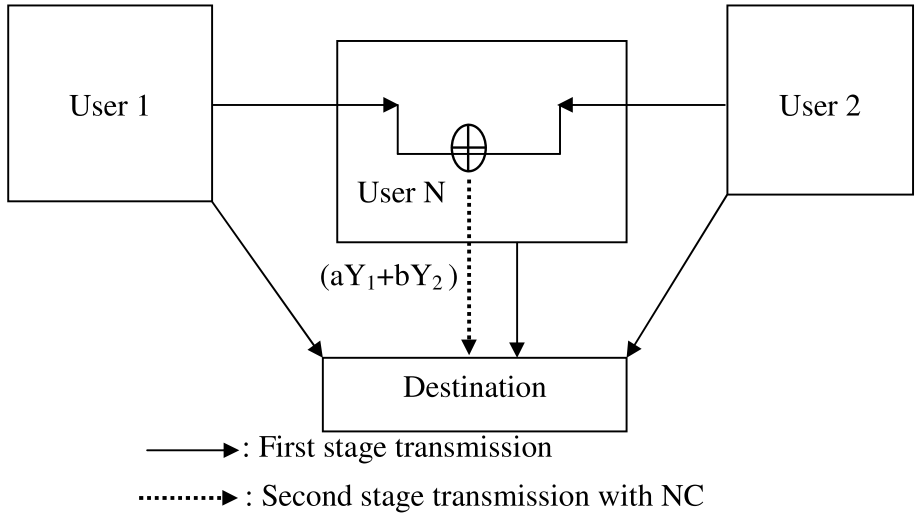


Figure 1.2: Two stages communication.

In the case of not receiving all packets at the destination, the second stage follows, which is the stage were NC is applied as shown in Figure 1.2.
In such case, each user relays the received neighbour's packets after combining its own packet to the received ones.

The destination in this case will be receiving the transmitted packets uncombined from the first stage at the first time slot. In the second time slot, the re-directed packets are received to help enabling the destination to receive the wanted packet correctly.

Based on above, the system does not perform NC in the first stage as it could resemble Amplify-and Forward (AF) relaying, where the BS simply amplifies the received signals before forwarding, or the relay performs Decode-and-Forward (DF), where the BS decodes the received signals, before performing NC on the reconstructions as shown in Figure 1.3 which shows how both systems work in the first stage.

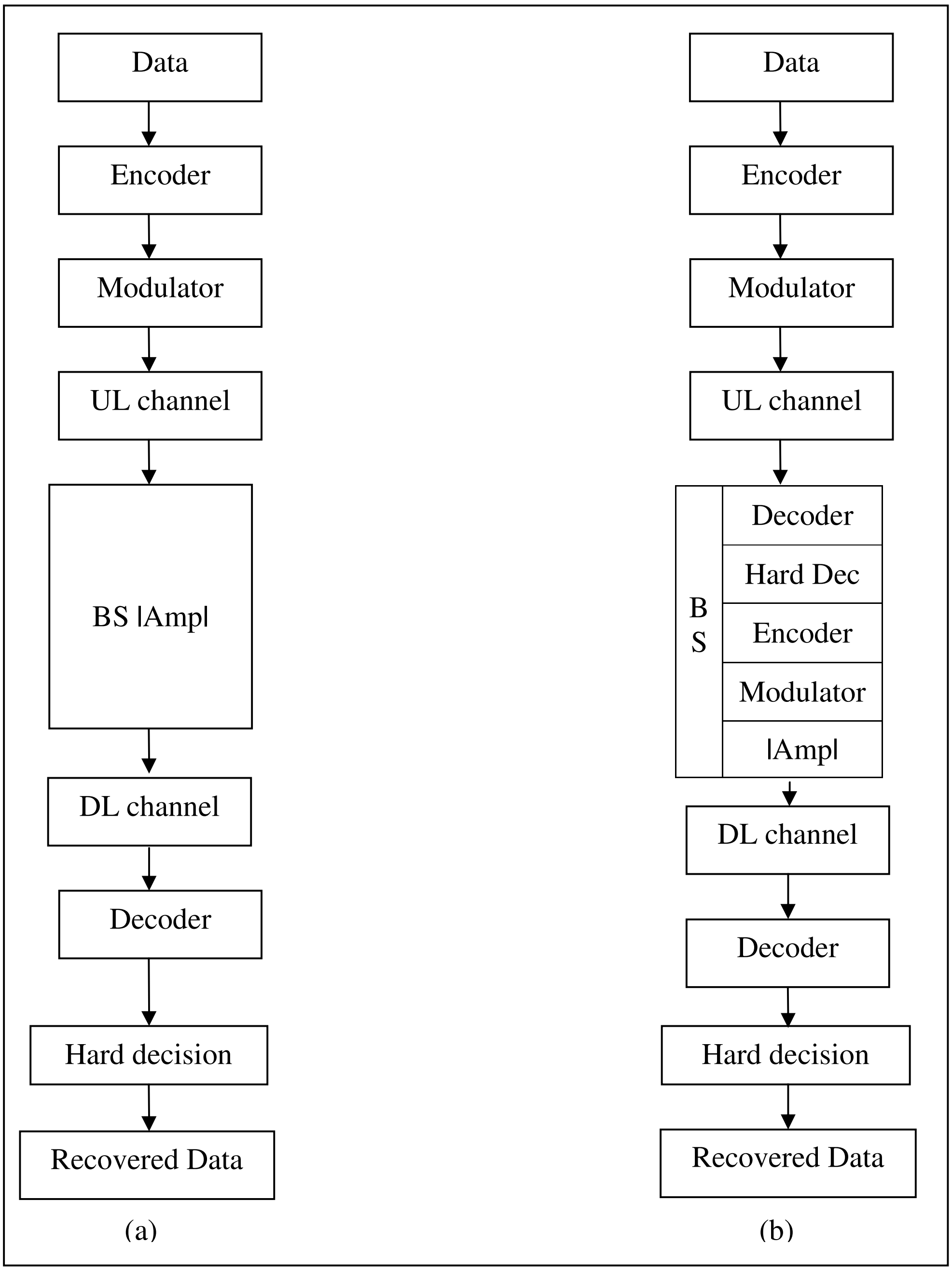


Figure 1.3: (a) Amplitude-and-forward base station, (b) Decode-and-forward base station.

In AF system, the BS just amplifies the received signal and then forwards it to be sent via the downlink channel. As we can see, in this system, BS does not exploit the broadcast nature of wireless radio links, as the BS is just used to amplify the signal.

In DF system, the BS performs in more complicated way, so, it decodes the received signal, retrieves the original signal by using hard decision, re-encodes, modulates and amplifies before sending the signal. This makes the system more complicated, and causes time delay to the system, but it gives better performance [8].

So, the BS decodes X_{En1} and X_{En2} separately, reconstructing a_{rec} and b_{rec}, where a_{rec} and b_{rec} are the reconstructed transmitted data a and b respectively. The reconstructed a_{rec} and b_{rec} are re-encoded and modulated to $X_{\text{ret-En1}}$ and $X_{\text{ret-En2}}$, respectively. Then the BS sends $X_{\text{ret-En2}}$ User 1 and $X_{\text{ret-En1}}$ to User 2.

Note that, the base station does not need to use the same codebook as the users. The signals received by Users 1 and 2, respectively are: $Y_{\mathrm{rec3}}=\mathrm{A}_{\mathrm{DFb}}\, X_{\text{ret-En2}} + Z_3$, and $Y_{\mathrm{rec4}} =\mathrm{A}_{\mathrm{DFb}}\, X_{\text{ret-En1}}+ Z_4$, where $\mathrm{A}_{\mathrm{DFb}}$ is the amplitude Y_{BS} for $\mathrm{DF_b}$ and $Z_3$is the additive white Gaussian noise.

Based on the fact we mentioned above, the BS in both $\mathrm{AF_b}$ and $\mathrm{DF_b}$ does not exploit the broadcast nature of wireless radio links, as the BS is just used to either amplify the signal or decode before forwarding.

In the second stage, the BS tends to exploit the broadcast nature of wireless radio links. This means there is no need to send data packets separately, as they can be added together and then sent as one data packet. At the receiving part, the received combined packet is used to retrieve the desired signal. Cooperative principle allows neighbours nodes to behave as a BS for the rest of nodes with or without applying NC, but by applying NC over the received packets at the BS results to reducing the communication load. Then, the user node tends to amplify the resulting signal or decode and re-encode before amplifying and then broadcast the resulting signal in term of one signal as (|Amp(YBS1+YBS2)|).

Figure 1.4 shows the system with applying NC over two users:

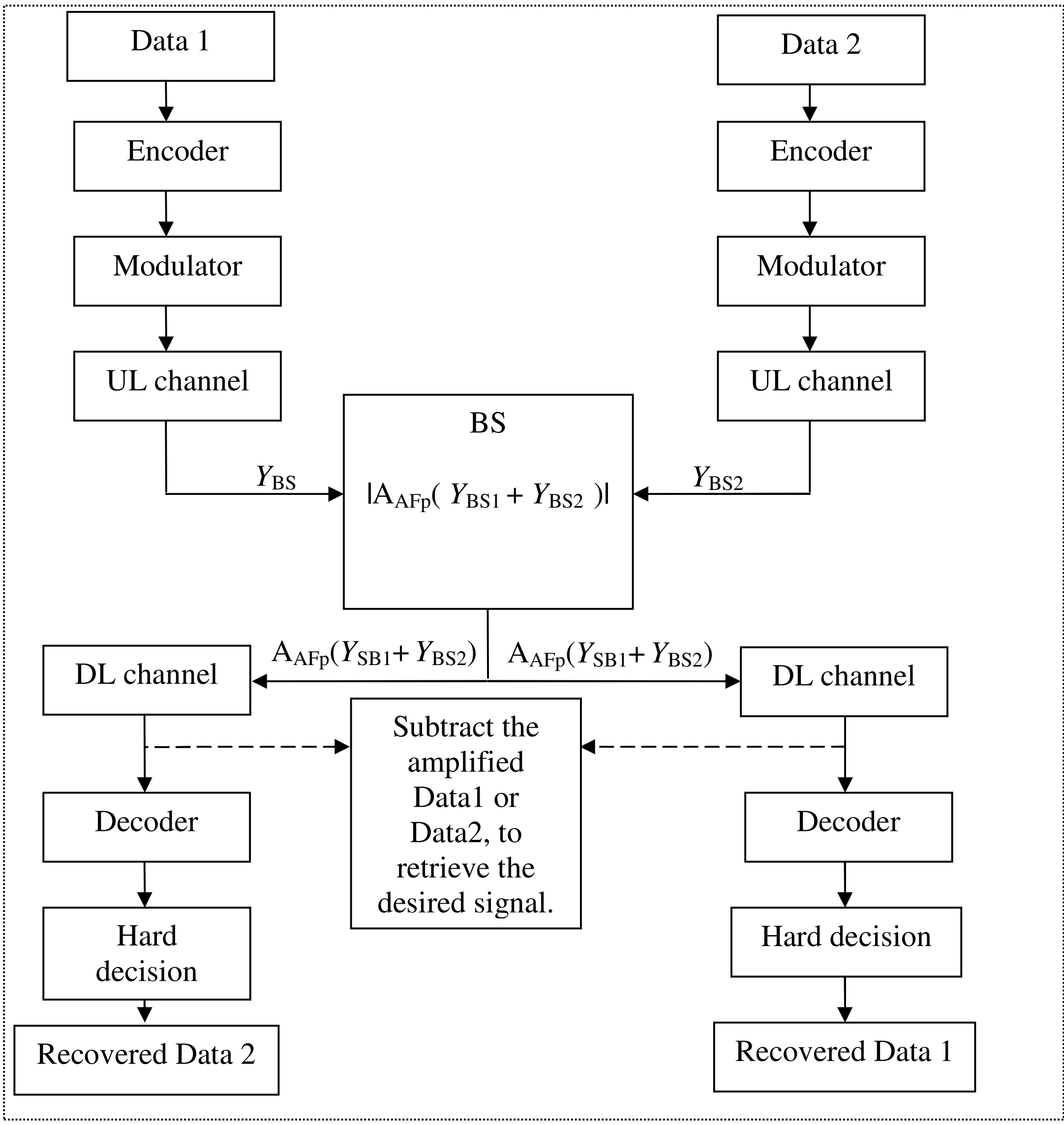


Figure 1.4: Amplitude and Forward system with Network Coding over two users.

According to Figure 1.4, the signals received by Users 1 and 2, respectively, are: Y_{rec3}=$A_{AFP}(Y_{BS1}+Y_{BS1}) + Z_3$ and Y_{rec4}=$A_{AFP}(Y_{BS1}+Y_{BS2}) + Z_4$; where the amplification coefficient of the proposed scheme is $A_{AFp} \geq 1$, and $Y_{SB1} = X_{En1} + Z_1$, similarly, $Y_{SB2} = X_{En1} + Z_2$. User 1 subtracts $A_{AFp}X_{En1}$ from Y_{rec3} yielding $X_{ret\text{-}En1}$= $A_{AFp}(X_{En2} + Z_1 + Z_2) + Z_3$, and then decodes it to reconstruct the message X_{ret2}. Similarly, User 2 subtracts AFpX_{En2} from Y_{rec4} yielding $X_{ret\text{-}En2}$= $A_{AFp}(X_{En1} + Z_1 + Z_2)$ + Z4, and reconstruct the message X_{ret1}. The following Figure 1.5 shows the Decode and Forward DF system with applying NC over two users:

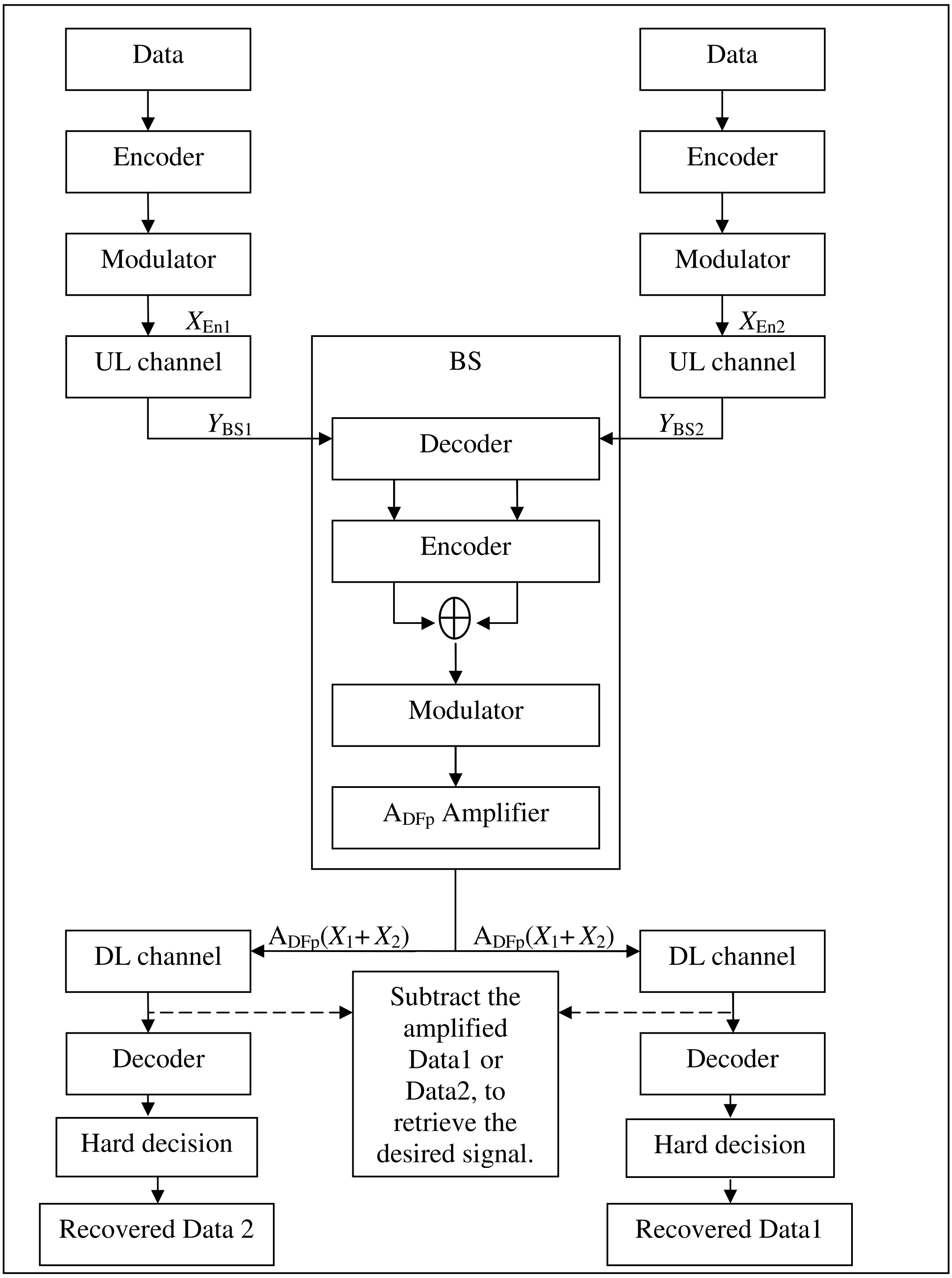


Figure 1.5: Decode and Forward system with Network Coding over two users.

As shown in Figure 1-5, the BS separately decodes Y_{BS1} and Y_{BS2} to reconstruct $X_{ret\text{-}En1}$ and $X_{ret\text{-}En1}$.Then, the reconstructed messages $X_{ret\text{-}En1}$ and $X_{ret\text{-}En2}$ are re-encoded and modulated to X_1 and X_2, respectively; X_1 and X_2 are summed and broadcast to the users. Thus, the signals received by Users 1 and 2, respectively, are:
$Y_{rec3} = A_{DFp}(X_1+X_2) + Z_3$ and $Y_{rec4} = A_{DFp}(X_1+X_2) + Z_4$: User 1 subtracts $A_{DFp}X_1$ from Y_{rec3} yielding $Y_{ret3}=A_{DFp}X_2+Z_3$, and then decodes it to reconstruct message X_{ret2}. Similarly, User 2 subtracts $A_{DFp}X_2$from Y_{rec4}, yielding $Y_{ret4}=A_{DFp}X_1+Z_4$ and reconstructs X_{ret1}. Taking into consideration that hard decision is used to convert the soft decision value to the retrieved data.

1.5 Forward Error Correction

Forward Error Correcting (FEC) codes are used at the receiver end to correct errors introduced by the channel, to negate the need for retransmissions, mainly in the systems where retransmissions are particularly expensive in term of performance. This can be performed at the receiver side by adding extra bits to the data packets, which called redundancy bits. The redundant bits are used to correct the expected errors within the packet to restore any corrupted packets at the destination. The redundant data is sometimes known as Error Correcting Code (ECC) and can be of different content depending on the kind of correction coding and FEC being used. The type of FEC determines the limit that ECC can restore from the original corrupted packet and in the case when there is a too large portion of the original packet is corrupted; then the ECC may fail to restore the packet properly [14].

Turbo codes and Low Density Parity Check (LDPC) are popular channel codes used at Physical Layer to correct noise and fading effects which are called Shannon limit approaching FEC codes. In the MAC/application layer, fountain and raptor codes are used to correct erasures packet loss. In this research, we used the convolutional FEC codes in the Physical Layer, that add redundant bits to the original message, of size k bits, resulting in an overall code size of n bits, with code rate of $R=k/n$.

1.6 Hybrid Automatic Repeat Request HARQ

Though HARQ is a technique was used in later 3G, it is extensively used in 4G networks. Moreover, HARQ is being used in some applications in Universal Mobile Telecommunications System (UMTS), WiMax and LTE.

HARQ is used widely in Release 10 3GPP standards, which is as a technique, combines FEC and ARQ techniques, which both are older technologies have been widely used for some time.

The HARQ protocol in the simplest form, takes certain aspects of both ARQ and FEC to allow them to work together effectively. In its most basic form, HARQ will add redundancy bits to its packets, usually less than FEC would, but it will also allow for retransmission in a similar manner to ARQ, i.e., HARQ protocol is able to correct smaller errors in packets than FEC does but requests fewer retransmissions than ARQ does. In such case, the retransmission is requested just when the packets have large errors, as the receiver in this case, is not able to restore the corruptions in the data to an acceptable level. Based on above, HARQ allows data to be transmitted more effectively by minimizing the inefficiency of ARQ and FEC.

In FEC, the addition of error correcting bits can, significantly increase the size of the data field in the packet, where as in HARQ, there is only need for a small amount of correction code, which reduces the size of the packets as well as reducing the complexity of the required EEC. HARQ is also more efficient than ARQ in term of reducing the amount of retransmissions which are required by repairing many packets with the redundancy bits, whereas in ARQ any error which is detected will automatically require retransmission, which can be very time consuming in some systems. There are also more sophisticated manners to increase HARQ‟s effectiveness. However, simple form of HARQ is particularly effective in poor signal conditions, this can mean significantly low levels of throughput in good signal conditions [15].

HARQ in more sophisticated form, attempts to only include FEC redundancy bits when they are necessary, which means sending the data block first and only if an error is detected; the FEC bits be sent. This reduces the average size of data packets being sent and keeps transmissions as efficient as possible [16].

To improve the efficiency for HARQ, different amount of the redundant data bits are sent if needed instead of a uniform amount each time, resulting to further reduction in average packet size. Any redundancy bits which are retransmitted are stored and used to restore erroneous data [17].

In some forms of HARQ a process known as 'soft combining' is applied to HARQ and can achieve significant affects on performance, where incorrectly received coded packet is stored at the receiver to combine it with the received re-transmitted packet [17] . The reason behinds this combination is the possibility to decode the two combined packets if they both are received with uncorrectable error rather than decoding them separately, as it may happen that the combination of the previously erroneously received transmissions gives us enough information to correctly decode.

Reference [18] shows the two main soft combining methods in HARQ as following:

- Chase combining: This combination is used when every retransmitted packet contains the same data and parity bits; basically, it is an additional repetition coding which requires extra energy. This is to enable the receivers to perform maximum-ratio combining to combine the received bits with the same bits from previous transmissions.
- Incremental redundancy: Unlike chase combining, incremental redundancy is used to provide the receiver with extra knowledge through the re-transmitted packet. In such case, the two transmitted packets are not identical, though they both have the same set of information bits. This is because the retransmission uses a different set of coded bits with different parity bits order which generated by puncturing the decoder output.

In [19], a comparison between ARQ and the mentioned HARQ types has been carried out, and found that the most effective method was sending a uniform FEC when necessary with 'code combining' applied to it. Sending different size of redundant bits when necessary was generally less effective than the uniform FEC redundancy with code combining; however, there is an exception in performance at SNRs.

In this research, we are applying NC to reduce the retransmission instead of HARQ over 4 G in both uplink and downlink transmissions.

1.7 Shannon Limit Approaching Forward Error Correction Codes

We use in this section the introduction which is useful to our research; mainly form [20] and [21], with [22] where Shannon introduced the research that uses modern fields of information theory and coding theory. Shannon's research presented a useful and explicit theoretical definition for the channel capacity C, which shows the information rate that could be transmitted reliably over the communication channel. Moreover, the formula for the wireless propagation channel capacity at a given bandwidth was provided and measured in bits/sec received. This formula is linearly proportional to the bandwidth and logarithmically proportional to the received signal-to-noise ratio.

The theorem proof shows that the coding scheme with acceptable small error probability is achievable when the transmission rate $\boldsymbol{R} \leq C$. In the case when $\boldsymbol{R} \geq C$, error free communication coding scheme could not be achieved.

Despite that Shannon gave such valuable theorem limits, but it does not show how to find the coding schemes and what complexity is raised when they are implemented.

The work in [23] and the work after, have developed many schemes for different applications to obtain the promised performance by Shannon or as close as possible to it at acceptable level of complexity.

Finite-field algebraic concepts have been used in other way of attempting to construct encoders and decoders which provide approaching performance to Shannon's channel capacity, which opened the gate for powerful error correction and detection in computers and storage devices which have major influences on the economics, particularly for wireless and other physical channels [24].

The first classes of Shannon-limit-approaching codes appeared in 1993 and named Turbo Codes by Berrou et al [25].

Turbo codes obtained performance less than 1.0 dB away from Capacity even with short constraint length. In addition to the Turbo codes, very long block lengths and soft decision iterative decoding has obtained this performance.

In [26], Low Density Parity check codes obtained another class which is close to Shannon limit codes, indeed, it reached a 0.0045 dB gap from Shannon-limit using 1/2 rate code of block length 10^7 at BER 10^{-6}.

Recently, [27] and [28] showed that the Turbo and LDPC decoding algorithms can both be derived as specific instances of one algorithm, which is called the *sum-product* algorithm.

1.7.1 Low Density Parity Check Codes

Low Density Parity Check (LDPC) was reintroduced Gallager [29] and [30], after being ignored for over three decades, until recently rediscovered and classified as another class of Shannon limit approaching codes [26] besides turbo codes Berrou, et al (1993) in [25]. LDPC codes are decoded with iterative decoding algorithms, such as the sum-product algorithm shown by [27], which achieves amazingly good error performance.

Gallager [30] and [31] Length n binary LDPC code over GF(2) is defined as a $J \times n$ null space parity check matrix **H** which satisfies the structure with ρ``ones" in each row, $\gamma$ ``ones" in each column, λ ``1-components" between any two rows or any two columns is either 0 or 1, and γ and ρ are small when compared with n. This means that the matrix **H** has constant ρ row and γ column weights. Moreover, setting λ to be either 0 or 1 and choosing γ and ρ to be small compared to n, enable the matrix **H** to not generate a LDPC with short cycles in the Tanner graph [32]. The Tanner graph is explained in [33] which shows that the Tanner graph has two vertices level; the variable vertices which represent the n code bits and the J vertices which is the J check equations that the code bits must have.

Figure 6.1 shows that a code bit vertex v_l is connected to a check sum s_j when only the **H** matrix entry j row and l column is “1”. Moreover, no more than one connection is allowed among the vertices.

$$\mathbf{H} = \begin{pmatrix} 1 & 1 & 0 & 1 \\ 1 & 0 & 1 & 1 \\ 0 & 1 & 0 & 1 \end{pmatrix}$$

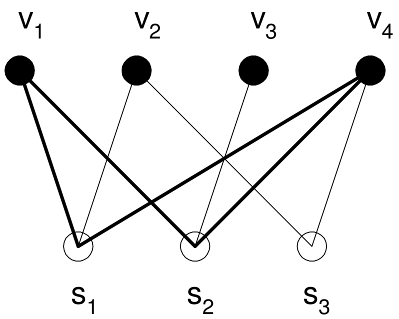


Figure 1.6: A binary matrix and its corresponding Tanner graph. A cycle of length 4 is emboldened.

A cycle in a graph of vertices and connection represents the sequence of connected edges which start from and end at the same certain vertex with no more than one vertex in the cycle [34]. The cycle always exists and not less than 4 in length, moreover, the shortest cycle in the Tanner graph is called ‘girth’.

1.7.2 Convolutional Code

The convolutional code is a type of FEC codes used at the physical layer, in which each *k*-bit information symbol encoded into an *n*-bit symbol, with *k*/*n* code rate and *k*-*n* redundancy bits, used for error detection and correction. Three integers are used to describe the convolutional codes, (*n,k,K*), where *K* is the constraint length; it represents the number of *k*-tuple stages required to drive a path in the trellis to the zero-state, which is practically used to control the redundancy [39].

Unlike traditional codes, such as block code, convolutional encoder has memory, so, the present output has a direct relation with the previous ones, according to the way the encoder and decoder are designed. Trellises offer an effective way to represent and summarize all the convolutional encoder’s output possibilities for given inputs and memory states.

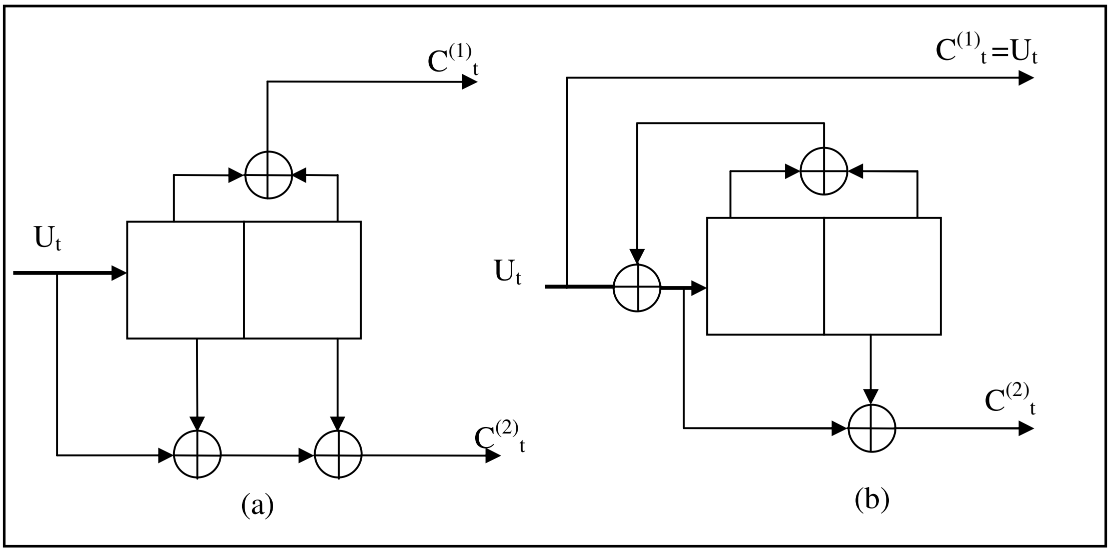


Figure 1.7: (a): Convolutional encoder and (b): Recursive Systematic Convolutional encoder.

Figure 1.7 and Figure 1.8 show an example for convolutional encoder and its trellis. Note that in the recursive Systematic convolutional encoder Figure 1.7 (b), the input is a part from the output.

"The Hamming distance for convolutional codes is referred to as its free distance, d_{free}. d_{free} is the minimum weight of a codeword that diverges from the all-zero path in the encoder standard trellis and returns to the all-zero path for the first time" page 13, [39], accordantly, " d_{free} for the encoder shown in Figure 1.7 (a) is 5" page 13, [39].

The convolutional codes can be decoded by using several algorithms. For relatively small values of *k*, the Viterbi algorithm is universally used as it provides maximum likelihood performance and is highly parallelizable.

Figure 1.8 (trellis), gives all the output possibilities, for the encoder input shown if Figure 1.7 (a) from the current state to the next state, and it shows how the present state depends in the previous ones, taking into consideration that the solid represents an input of 0 to the encoder and the dashed represents an input of 1 in the Figure 1.8 [39].

The trellis is continuous over time *t* as long as there are inputs being entered into the encoder.

One other important note is the number of the trellis states depends on the number of the memory registers in the encoder. So, for an encoder with M_r memory registers, the number of the trellis states is 2^{Mr}.

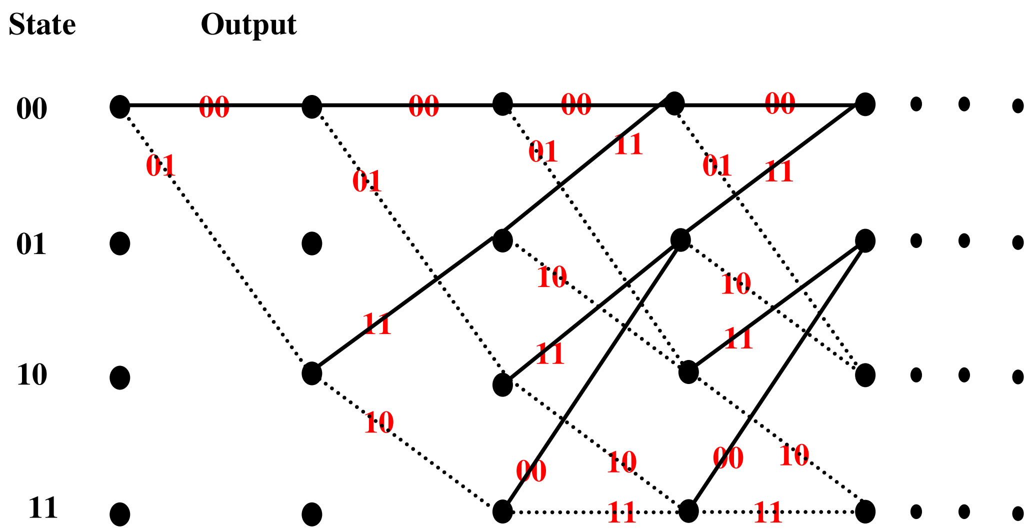


Figure 1.8: Trellis of convolutional code for the encoder in Figure 1.7 (a).

1.7.3 Partial Unit Memory (PUM) Code

Partial Unit Memory codes were introduced by Lauer in 1979 [40]. PUM codes are multiple-input convolutional codes, which gives the optimal Maximum free distance versus the data rate and number of memory units and input bits.

Their main advantages for PUM code over classical convolutional codes are the reduced number of states in the trellis diagram for the same number of input encoder bits, and a larger free distance for the same trellis state complexity.

Because the number of memory units is more than zero and less than the number of the input bits, PUM codes holds an intermediate position between block (zero memory units) and convolutional codes (more memory units than input bits), as shown in [39] and [41], where M is the memory size in number of bits and k the number of input bits to the encoder.

Table 1.1: Relative memory complexity of codes

Block Code	PUM Code	UM Code	Convolutional Code
M = 0	M < k	M = k	M > k

As a result for the intermediate memory structure, PUM code has the properties of both block and convolutional codes enabling it to be suitable for many practical systems. Moreover, the non-binary inputs and outputs natural of PUM code enhances a larger flexibility for the generation of rate-compatible and adaptive codes [39] and [41].

A PUM code [39] is characterized by four parameters (*n; k; μ; dfree*), where n is the codeword length, k is the number of information bits to be encoded, μ is the memory (i.e., the number of bits in the shift register), and *dfree* is the minimum (free) distance between any two code sequences. The output word *Ct* of an (*n; k; μ; dfree*) PUM code is a function of the current input word of k information bits, *ut,* and a fraction μ (where $\mu <$ k) of the previous input word ut¡1. Memory μ determines the state complexity of the code trellis diagram - the lower the μ the lower the decoding complexity. A convolutional code trellis is made up of 2^{μ} states with 2^{k} branches leaving and entering each state. For PUM codes, since $k > \mu$, there are $2^{\mu-k}$ parallel branches between any two states in the trellis [39]. A number of non-systematic generator matrices for PUM codes have been reported [42] with distance properties better than those of multi-memory convolutional codes of the same state complexity.

1.8 Random Linear Code RLC

Random Linear Codes (RLC) is a class of erasure protection codes which is suitable for short message length scenarios. Indeed, RLC is considered as promising rateless coding solutions. The main advantage RLC provides is the performance that is regarded as close to optimal FEC solution over erasure channels. Moreover, RLC is the main gate to fully understand NC concepts [43]. However, RLC suffers from the high decoding complexity mainly when increasing the length of the source's messages as decoding complexity of Gaussian elimination decoding is significantly increases with the length of the source messages. The relation between NC and RLC makes this application to be helpful in the most modern applications, mainly to increase the coverage and the throughput such as in today's cellular systems that consider multi-hop topologies using relays, as well as device-to-device data exchange. When RLC is applied over a source message, it produces encoded symbols containing random linear combinations of source symbols with coefficients randomly selected from a given finite field. Moreover, RLC can be applied over the relay with NC framework in one or multiple communication session. On the other hand, as a packet-level FEC solution, RLC is simple to implement and performs to be an optimal erasure codes for sufficiently large finite field used which makes RLC to be such attractive universal FEC/network coding solution for emerging wireless communication systems [44].

1.9 Fountain Codes

The digital fountain code is such efficient way to negate the retransmission. This can be performed by inserting a stream of distinct encoding packets into the network, which used by the receiver to reconstruct the source data. The injected streams of packets are integrally reconstructed by the source, from any subset of the encoding packets with equal length to the source data [45]. Digital Fountain is implemented by directly using an erasure code that takes source data with k packets length, and produces enough encoding packets to meet user demand. In the standard erasure codes, such as Reed-Solomon erasure codes, receiving just k encoded packets is enough for the decoder to reconstruct the original source data at the receiving side. However, typical erasure codes stretch the k packets file into n encoding packets, where the ratio n/k is called the stretch j factor and it is the internal parameter of an erasure code. Reasonable approximation for j is important and it is set n to be a multiple of k, then repeatedly cycle through transmission of the n encoding packets. One important fact should be considered when designing the digital fountain codes, which is, under sufficiently high loss rates, the receiver may not receive k out of n packets in one cycle; moreover, the receiver may receive useless duplicate transmissions before reconstructing the source data, which decreases the channel efficiency. However, experimental results show that such small multiple of k such as $n = 2k$, even under very high loss rates, the inefficiency is not large.The encoding and decoding processing times is such important factor when choosing the erasure code, for example, standard Reed-Solomon erasure codes have such long processing times, that is why they are prohibitive even for moderate values of k and n [46]. The Tornado Codes [47] is such better alternative as it is considered as fast but it requires a higher number of packets than *k* of the transmitted packets to reconstruct the source data. In general, choosing one code than the other depends on the trade off associated with the code, but the strength factor is considered as the main factor.

1.10 Raptor Codes

Raptor codes where introduced in [48] as the most powerful class of sparse binary rateless codes. The reason behinds this, is the flexibility to generate any desired number of encoded symbols from the source symbols which is a rateless property. The original source symbols can be decoded as long as the number of the generated symbols is at least equal or slightly exceeds the number of source symbols. The encoding and decoding algorithms are described in [49]. For source block size k and the number of received symbols n, the failure probability of Raptor code for $n \geq k$ decreases exponentially with increasing the number of received symbols. For roughly 12 additional symbols the failure probability is 0.1%, whereas for about 24 additional symbols the failure probability reduces to 0.0001% [50].

The digital fountain code is a very efficient way to negate the retransmission. This can be performed by inserting a stream of distinct encoding packets into the network, which used by the receiver to reconstruct the source data. The injected streams of packets are integrally reconstructed by the source, from any subset of the encoding packets with equal length to the source data [45]. Digital Fountain is implemented by directly using an erasure code that takes source data with k packets length, and produces enough encoding packets to meet user demand. In the standard erasure codes, such as Reed-Solomon erasure codes, receiving just k encoded packets is enough for the decoder to reconstruct the original source data at the receiving side. However, typical erasure codes stretch the k packets file into n encoding packets, where the ratio n/k is called the stretch j factor and it is the internal parameter of an erasure code. Reasonable approximation for j is important and is set n to be a multiple of k, and then repeatedly cycle through transmission of the n encoding packets.

One important fact should be considered when designing the digital fountain codes, which is, under sufficiently high loss rates, the receiver may not receive k out of n

packets in one cycle; moreover, the receiver may receive useless duplicate transmissions before reconstructing the source data, which decreases the channel efficiency. However, experimental results show that such small multiple of k such as $n = 2k$, even under very high loss rates, the inefficiency is not large. The encoding and decoding processing time is a very important factor when choosing the erasure code. For example, standard Reed-Solomon erasure codes have such long processing times, which is the reason they are prohibitive even for moderate values of k and n [46]. The Tornado Codes [47] are such better alternative that they are considered as fast but they require a higher number of packets than k of the transmitted packets to reconstruct the source data. In general, choosing one code than the other depends on the trade off associated with the code, but the strength factor is considered as the main factor.

To enhance the Video data transmission, Unequal Error Protection (UEP) is used, which divides the video stream according to its reconstruction priority, resulting in to dividing the data into unequal error protection, i. e., the more important division to reconstruct the video data, the better error correction is used. H.264/AVC provides a very low-cost data partitioning (DP) of video data based on its importance [43].

1.11 Wireless Sensor Network

WSN is spatially distributed autonomous devices using sensors (or sensor nodes) to cooperatively monitor physical or environmental conditions at different locations.

1.11.1 Sensor

The **sensor** is the main part in this technique which is identified as a device which converts measured quantities into a readable signal at the reception side.

1.11.2 Sensor Node

The **sensor node** is a node in a wireless sensor network that is capable of performing some processing, gathering sensory information and communicating with other connected nodes in the network. The typical architecture of the sensor node is shown in Figure 1.9:

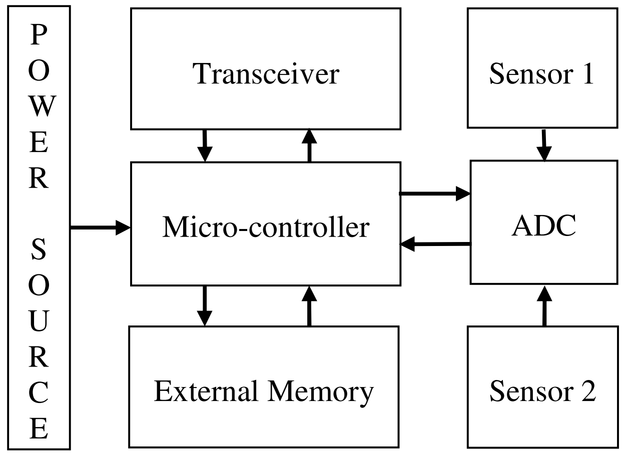


Figure 1.9: Typical architecture of sensor node.

The sensor's specifications determine the most important parameters for WSN device, such as the size, which varies from sand's size to the size of a shoes' box such as ZegBee transceiver board in WSN.

There are many types of the sensors, such as, thermal, electromagnetic, mechanical, chemical, optical radiation, ionizing radiation, and others.

1.11.3 WSN Characteristic

Unique characteristics of a WSN include:

- Limited power they can harvest or store.
- Ability to withstand harsh environmental conditions
- Ability to cope with node failures
- Mobility of nodes
- Dynamic network topology
- Communication failures
- Heterogeneity of nodes
- Large scale of deployment
- Unattended operation

To make the sensor nodes more understandable, we can imagine them as very small and basic computers, so, they have very basic components and interfaces. The reason we can compare the sensor nodes with computers, is because they usually consist of a processing unit with limited computational power and limited memory, sensors (including specific conditioning circuitry), a communication device (usually radio transceivers or alternatively optical), and a power source usually a battery [51].

1.11.4 Wireless Sensor Network Applications

WSN applications are ranging from battlefield surveillance and medical care to environmental monitoring. So, recent advances in sensor can be used in various ways:

Monitoring Space: Environmental and habitat monitoring, precision agriculture, indoor climate control, surveillance and intelligent alarms.

Monitoring Objects: Such as, structural monitoring, condition-based equipment maintenance and medical diagnostics.

Monitoring Interactions: Between objects and between objects and their environment, e.g. wildlife habitats, disaster management, emergency response, healthcare and manufacturing process flow.

1.11.5 Wireless Sensor Network Challenges

WSN faces number of challenges, we are listing the challenges that we are we are motivated to overcome.

- ***Energy Constrained Nodes:*** WSN nodes participating in the network are power limited, as they rely on batteries for power.
- ***Capacity Constraints:*** WSN has significantly lower capacity than wired links and hence congestion is more problematic.
- ***Mobility:*** WSN could be used for mobility applications with changeable speeds, such as monitoring moving animals. So, the Quality of Service (QoS) must take this aspect into consideration.
- ***Quality of Service Level:***

  The effective QoS required is subject to change mainly due to limited transmission power, user mobility and changeable channel conditions.

On the top of the above challenges, there are some important factors that need to be considered when designing a WSN which are: minimum delivery latency, higher probability of packet delivery and adaptability. Therefore, the design of an efficient and reliable broadcasting scheme and QoS support for such applications is a major challenge.

1.12 Long Term Evolution Advanced (LTE-A)

LTE-A is the project name for the Third Generation Partnership Project (3GPP) which aims to develop this standard to the Firth Generation of mobile networks (4G).

The most important features for the 4G is the capability of worldwide roaming and enhancing peak data rates to support advanced services and applications (100 Mbit/s for high and 1 Gbit/S for low mobility) were established as targets for our application example in this research. Based on above, the 4G networks almost certainly characterized by much higher data rates than the previous generation [52].

1.12.1 Coordinated Multi-Point

Coordinated Multi-Point (CoMP) is a technology which sends and receives signals from multiple sectors or cells to a located User End (UE) via coordinating the transmission among the multiple cells, to reduce the interference from other cells. As a result, the desired signal power can be increased [53].

To perform CoMP transmission and reception, orthogonalization on the uplink and downlink intracell/inter-cell is implemented in LTE Release 8 to meeting the requirements of capacity and cell-edge user throughput. Both the uplink and the downlink are simultaneously connected to UE in the frequency domain orthogonalizely. Moreover, the downlink is connected in the code domain as well, using cyclic shift and block spreading. It is possible to assign different frequency ranges for cell-edge UEs to control interference between cells semi-statically, but this is done based on randomization in LTE Rel. 8 [53].

1.12.1.1 CoMP Implementation

CoMP technology is implemented in two different ways which are autonomous distributed control based on an independent eNode B configuration, and centralized control based on Remote Radio Equipment (RRE) as shown in Figure 1.10 [54]:

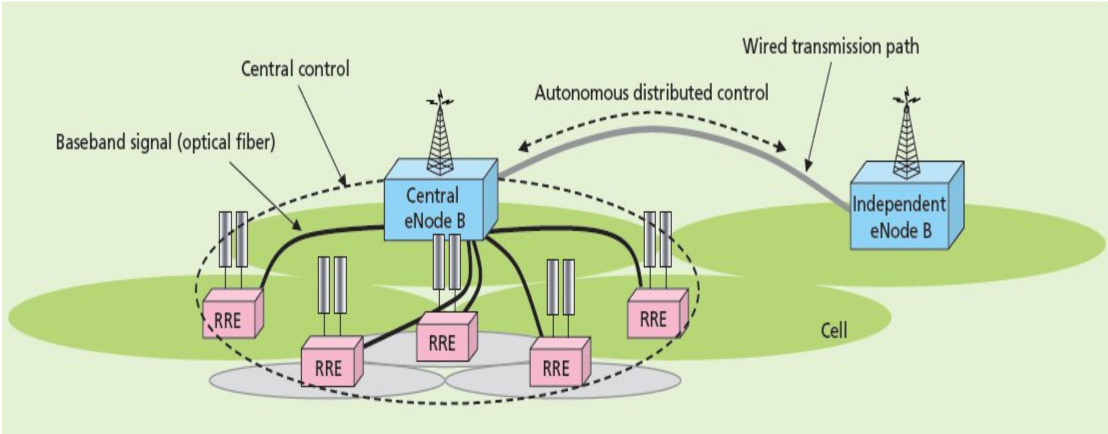


Figure 1.10: Central control and Autonomous distributed control Coordinated Multi-Point implementations.

In the first configuration, a signaling over wired transmission path is used to coordinate among cells, which can be performed with a regular cell configuration. However, increasing the signaling speed or perform high-speed signaling via UE is needed to overcome the signaling delay and overhead issues,

To decrease the effect of the signaling delay and overhead, RRE configurations is used by connecting multiple RREs via an optical fiber carrying a baseband signal between cells and the central eNode B. In this configuration, the central eNode B performs the baseband signal processing and control to control the radio resources between the cells at the central eNode B directly. To control high speed radio resources between cells, high capacity optical fiber is required.

The processing load on the central eNode B increases when the number of RRE increases, resulting to limitation in applying this configuration, which is the reason, it is important to appropriately use both distributed control based on independent eNode B configurations and centralized control based on RRE configurations in the preparation for LTE-Advanced.

1.12.1.2 Downlink Coordinated Multi-Point Transmission

Downlink CoMP transmission can be divided to Coordinated Scheduling/Coordinated Beamforming (CS/CB), and joint processing as shown in Figure 1.11 bellow [54]:

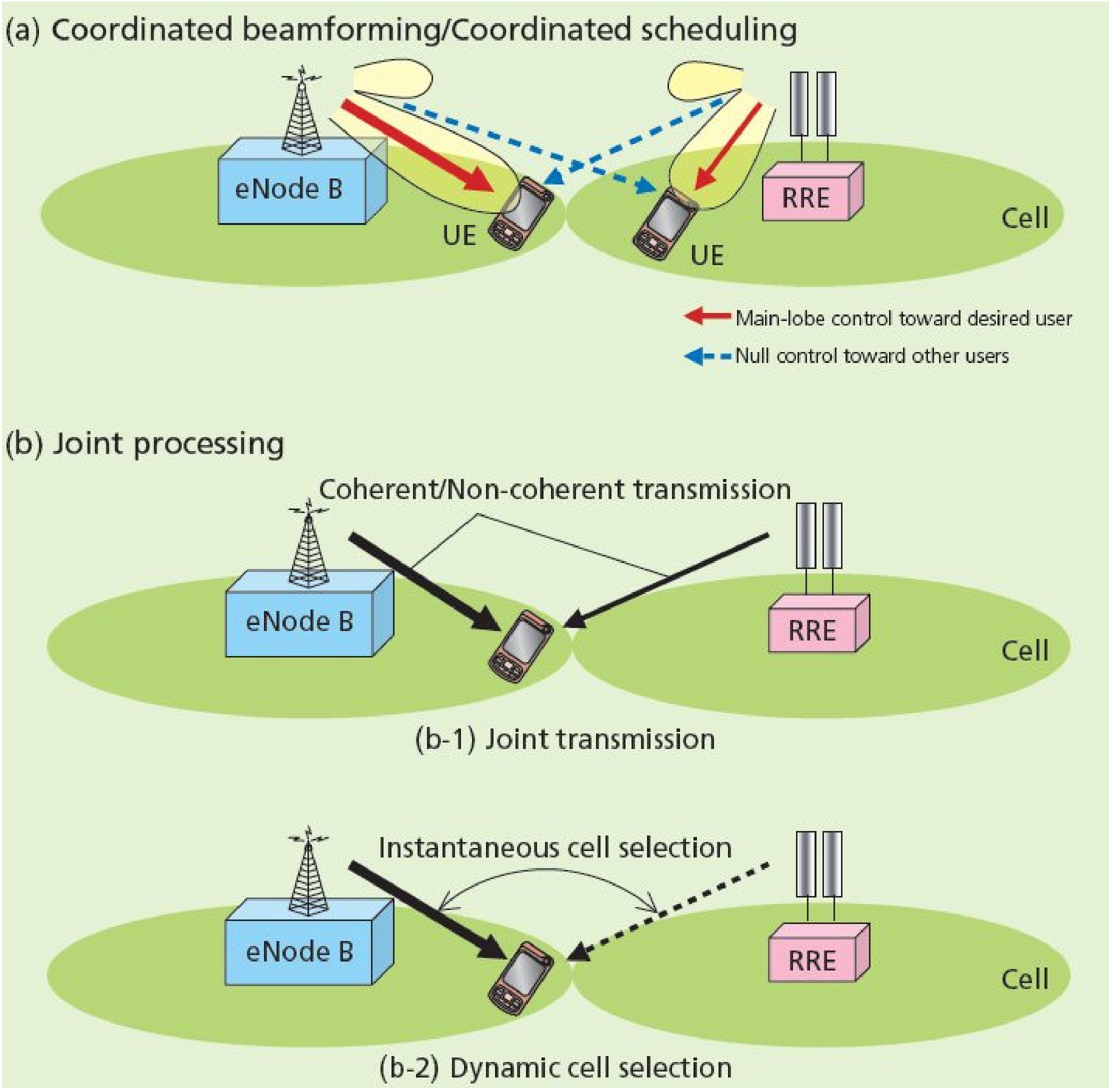


Figure 1.11: Downlink Coordinated Multi-Point.

Figure 1.11 (a) shows CS/CB when transmitting a given subframe from one cell to a located UE. In such case, coordinated beamforming and scheduling is performed between cells to reduce the other cells interference. In the case of joint processing, as shown in Figure 1.11 (b-1) and Figure 1.11 (b-2), a joint transmission by multiple cells to a located UE is utilized at the same time using the same time and frequency radio resources, and dynamic cell selection. The joint transmission is performed by coherent and non-coherent. In the coherent transmission, pre-coding between cells is implemented using in-phase combining at the receiver. In non-coherent transmission, soft-combining reception of the OFDM signal is used. Both methods are being studied.

In the *uplink multi-cell reception*, multiple cells can receive a combined signal from a UE regardless whether the UE is aware or not of the multi-cell reception causing little impact on the radio interface specifications.

1.12.2 Cooperation and Coordinated Multi-Point

The 4G networks established a fully heterogeneous network enabling different devices in nodes and made in a cluster to share the communication [55]. This important development improves the performance and reliability of the entire cluster.

Cooperation in the 4G has been gaining significant attention recently as more heterogeneous nodes are added to the environment; this high pace in the number of wireless devices makes it more common to have a number of devices grouped within the proximity of each other, which is known 'cluster of users'. All of the devices within this cluster can cooperate with each other and with other transmitting devices; this is beside the fact that each device is also connected to the cellular network. This means that the devices in a cluster will have "better communication performance in terms of, bandwidth usage, end system complexity and energy efficiency" [55].

As a result, the 4G Coordinated Multi-Point (CoMP) cooperation between cellular signals among devices is a promising direction for improvements.

Beside the cooperation ability, cooperative CoMP in the network cluster provides a range of different capabilities, as the cluster connects different kinds of devices for example, computers, mobile phones, modern cars and more. These devices can then share functionality and ordinarily provide a better service throughout the cluster [55].

In this research work, we use the key technology for clustering in 4G systems which is CoMP transmission and reception to increase the coverage and capacity of a cellular network by implementing cooperative communication between nodes in a cluster [56].

Uplink transmission CoMP works differently than the downlink. Moreover, it depends on the position of the mobile nodes in the cells and the type of scheduling algorithm which is applied to the network.

The CoMP is particularly effective in the uplink if a device is on the cell edge and can possibly receive signal from multiple cell sites, as this would allow signals from different cell sites to be scheduled effectively and the device would be able to make use of multiple signal sources. This would result in a significant improvement to the link performance in term of coverage area and capacity [57].

Using NC in the cell together with a BS is proposed in this work to extend the benefit from the cluster CoMP nodes, even if not located at the edge of the cell as will be illustrated later on, resulting to a significant improvement to the edge throughput, which is one of the most important targets for this research.

The cooperative CoMP has different techniques such are inter-site CoMP, intra-site CoMP and intra-eNB CoMP, full details for benefits and applications are shown in [58].

1.13 Cooperative Networking over LTE-A

The development of 4G mobile systems relies on the IMT-Advanced (International Mobile Telecommunications Advanced) requirements for 4G standards. Moreover, Intrusion Tolerance by Unpredictable Adaptation (ITU-A) defined that these requirements must be met to establish a truly 4G wireless system.

Recently, there are several researches to upgrade the 3GPP Release 8 and Release 10, to obtain the modern technology which can be classified as 4G. Indeed, LTE-Advanced (LTE-A) is considered to be the strongest candidate to become the first commercially deployable option in 4G.

These researches rely on the fact that, the key technology provided by 4G system for clustering is CoMP transmission and reception. CoMP technique supports the cooperative communication between the nodes in the cluster, which increases the coverage and capacity of the cellular network. This performance improvement is solely obtained by utilizing CoMP more of the cellular spectrum which is available to it than non-cooperative networks [56].

Based on above, we work to improve LTE which is classified as 3G, to obtain LTE-A which can be classified as a 4G by exploiting CoMP cooperative technique with NC.

In order to reach the standards that are accepted as a 4G, LTE-A has to pass the standardized level as part of Release 10 of 3GPP‟s standards [52]. And this means LTE-A must satisfy certain performance specifications. These are:

- Peak data rate of 1Gbit/s in the downlink.
- Peak data rate of 500Mbit/s in the uplink.

In addition to the peak data rates it must also meet other technical and operational specifications shown in [59].

In [60] HARQ-random NC has been implemented for a packet erasure channel for real-time media broadcast over singlehop wireless networks.

Unlike our stationary Femto Relay(s) [61] introduces applying NC with HARQ over a mobile relay systems using a 1/2 recursive systematic convolution code over BPSK. This work shows that even in such scenario, implementing NC with HARQ scheme provides an improvement in the throughput performance compared to existing relay-based HARQ (relay-HARQ) schemes.

In [62] NC with cooperative HARQ protocol was presented over two users connected to the base station via a relay node. In this scenario, the relay node responds to the repeat request and processes signal superposition modulation over the physical layer before transmitting to the base station.

Accordingly, Unlike [60], [61] and [62], our aim in this research work is to demonstrate how NC can be deterministically applied to the transmission process of a LTE-A system to reduce the ARQ significantly and hence improves the reliability and increases the throughput via using CoMP technique, which is implemented to increase the throughput on the uplink and downlink to satisfy the 4G specifications.

Table 1.2 illustrates LTE-A required specifications compared with 3GPP Release 8 LTE in terms of peak data rates and peak spectral efficiency.

Table 1.2 specification requirements for LTE-A [63].

	Down Link /Up Link	Antenna Configuration	Rel. 8 LTE achievement	LTE-Advanced
Peak data rate	Down Link	--------	300 Mb/s	1 Gb/s
	Up Link	--------	75 Mb/s	500 Mb/s
Peak spectrum efficiency [b/s/Hz]	Down Link	--------	15	30
	Up Link	--------	3.75	15
Capacity [b/s/Hz/cell]	Down Link	2×2	1.69	2.4
		4×2	1.87	2.6
		4×4	2.67	3.7
	Up Link	1×2	0.74	1.2
		2×4	--------	2
Cell edge user throughput [b/s/Hz/cell/user]	Down Link	2×2	0.05	0.07
		4×2	0.06	0.09
		4×4	0.08	0.12
	Up Link	1×2	0.024	0.04
		2×4	--------	0.07

Table 1.2 shows that there is an improvement in LTE-A on every aspect considered. However, the performance improvement rate is not the same for all specifications, i. e., some improving rate is more than others. For example, the peak data rate on the downlink, LTE-Advanced gives more than 200% of a performance improvement than LTE while the uplink improvement is at an even higher rate than that. On the other hand, the cell edge user throughput improvement for LTE-A is just 50% over Release 8 LTE and never goes above this, which is much less than other aspects. This is the main reason why we focus our research over this aspect to improve, i.e., we apply NC over CoMP and HARQ to improve the cell edge user throughput using the advantage of the

cooperative ability supported by CoMP over a cluster of users. Our results are compared to the already established results by the 3GPP, which is set as our benchmark. So, in the successful application of this project, the figures should act as guidelines for this benchmark to show the improvement has been obtained in the final adaption of the network configuration.
To understand LTE-Advanced, important concepts are shown below.

1.13.1 Release 10 LTE-Advanced

In late 2009, Release 10 of 3GPP standards was published, and therefore LTE Advanced was proposed as a candidate for IMT Advanced. The term advanced is used to identify the enhancements to LTE in Release 10 of the standards, leading to LTE-A [64], as a result, LTE-A were used to show high performance levels compared to any 3G technology, i.e., all specifications of LTE-Advanced are an improvement to LTE. The specifications which are of direct interest in our research can be deduced from Table 1.2 as bellow:

- Peak Data Rate: (uplink) - 500 Mbps.
- Spectrum efficiency: 3 times greater than LTE.
- Peak spectrum efficiency: (uplink) - 15 bps/Hz.
- Cell edge user throughput to be twice that of LTE (50% improvement at maximum level).
- Average user throughput to be 3 times that of LTE.

As mentioned in the previous Section, our interest is focused at the ability of cooperating that a 4G LTE-Advanced network support over a cluster of users. Moreover, our research aims to improve the edge cell throughput as it is the less progressed aspect in LTE-A over LTE.

1.14 Erasure Channel Modelling

The error free communication of video data over multi-hop wireless networks is a challenging research problem. Random linear codes have had renewed interest fostered by the multi-hop and multi-interface radio receivers to help in improving the communication quality. Indeed, LRC is able to generate as many symbols as needed to decode the transmitted set symbols, resulting to a fountain coding property were a small added fraction of the number of the transmitted symbols are generated randomly to decode the transmitted set of symbols. In this study, the descriptions are created using the encoding features of slicing and data partitioning.

The used channels and transmission model over LTE-A are shown in Section 2.5 and the subsection there in.

In our work over the physical layer, AWGN channel is used with the white additive Gaussian noise, and in the Media Access Control (MAC) layer, the erasure channel is used with simple erasure probability and with its Rayleigh block-fading channel PER to represent the erasure probability over the MAC layer.

1.15 Research Motivation

This research is directed to applying Cooperative NC over different wireless communication applications. In addition, this research investigates the results of applying NC over the physical layer and the upper layers. Accordingly, we started with applying NC over physical layer motivated by the desire to explore the BER penalty we are paying for combining the packets in the physical layer. Moreover, applying the NC over the physical layer shows the benefits from applying the decoding in the Base Station instead of just forwarding the packets after applying NC over them.

The success NC showed in the physical layer, with the proposed FFC turbo code, mainly with DF system, encouraged our research to apply NC over modern applications in wireless communication. In fact, we went even beyond the physical layer to follow the transmitted data through all layers transmission. So, motivated to extend applying NC over other applications and layers, where, as we believe, NC has other positive influences than the saving in bandwidth.

Power efficient protocols design with apply NC were behind the motivation to choose WSN as a research application.

WSN is used mainly for provisioning applications such as monitoring space, objects and interactions, has received a significant attentions among practical researchers mainly due to the increasing demand on this service among users [1] and references there in.

Moreover, the research over WSN is also motivated by the other challenges mentioned in 1.10.5, as in general, wireless nodes have limited resources like capacity and battery power. In WSN, one of the key issues is how to broadcast packets efficiently even in mobility cases. Some of the important factors that need to be considered in designing a WSN are: minimum delivery latency, higher probability of packet delivery, energy efficiency and adaptability. Therefore, the design of an efficient and reliable broadcasting scheme and QoS support for such applications is a major challenge, which fuelled our motivation.

In fact, applying cooperative NC over WSN allows us to investigate the lossy channel in more details with erasure channel and Finite State Markov Chain, resulting to several protocols. These protocols showed positive attitude towards PER, in fact, they improved the PER significantly in some system's design.

Finally, after revealing Release 10 in LTE-A, and the CoMP and HARQ protocols 4G supports, we directed our cooperative NC applications to be applied over this area. To enable LTE-A aspects to be the first true 4G, CoMP protocol allows cooperation between nodes, this cooperation between nodes was the key to open this application's research, aiming to increase the coverage range and data rate in a serious attempt to achieve 4G.

1.16 Research Contribution

The contributions of this thesis are directed toward tackling the challenges mentioned in 1.10.5 for WSN, i.e., power efficient protocols, power efficient FEC codes, and EST. On the other hand, the benefit from using NC mentioned in Section 1.2 extends our contribution to be extended over different layers, such as MAC and for different applications, such as LET-A.

Contributions can be divided according to either the layer we force our contribution over, or according to the channel used, such as FSMC and erasure channel.

1.16.1 Network Coding over Physical Layer

In this contribution, we propose two practical power- and bandwidth-efficient systems based on AF and DF schemes to address the problem of information exchange via a relay. The key idea is to channel encode each source's message by using a high-performance non-binary turbo code based on Partial Unit Memory (PUM) codes to enhance the bit-error-rate performance, then reduce the energy consumption and increase spectrum efficiency by using NC to combine individual nodes' messages at the relay. Two simple and low complexity physical layer NC schemes are proposed based on spatial and temporal combinations of received source messages at the relay. We also present the theoretical limits and numerical analysis of the proposed schemes. Simulation results under Additive White Gaussian Noise, confirm that the proposed schemes achieve significant bandwidth savings and fewer transmissions over the benchmark systems which do not resort to NC. Capacity theoretical limits and behaviour at Signal to Noise Ratio (SNR) for the proposed schemes is shown. The contribution also proposes a cooperative strategy that is useful when insufficient combined messages are not received at a node to recover the desired source messages which enables the system to retrieve all packets with significantly fewer retransmission request messages. Finally, out of revealing PUMTC behaviour, our contribution went to the point of illustrating the influence of increasing the amplification factor and decoding iteration.

1.16.2 Network Coding over Erasure Channel

We design and analyze a complete system that enables cooperation among multiple users through the use of NC in erasure channel. Starting with a simple erasure channel where all users to users probabilities are the same, and all users to destination probabilities are the same. Then, we then assumed that the erasure probability between the users and the destination varies according to the transmission distance. FSMC has been modeled as distance and time varying application, which makes the erasure channel to have different transmission probabilities for different distances at different transmission times. Full analysis for the probability of recovering all users' messages at a common destination, has been shown. We derive analytical expressions for probability of packet error at the destination for several different low complexity protocols with and without binary NC.

Applying NC over the packets received by the relay in the application layer, shows significant improvement in term of error probability and reduces the number of the requested repetition when one or more packets are not received after the first stage.

1.16.3 Network Coding with Node Cooperation

When NC is applied, we have the ability to design different protocols. These protocols are designed mainly to be power efficient. However, when applying cooperative with the NC, the benefits for these protocols go beyond of being just power efficient protocols. Indeed, cooperative allows the system to retrieve the lost packets instead of retransmitting. This improves the packet error probability for the proposed protocols, improves the data rate, the channel bandwidth and the reliability.

1.16.4 Network Coding LTE-A

In LTE-A, NC can be applied over the Femto relay(s), so, the Femto relays tend to combine the received packets than just forwarding them to the users.

Under such scenario, our contribution is aimed at finding the optimum protocol that can be applied over the Femto relays, in term of power efficient and connection.

1.16.5 Video Broadcasting to Heterogeneous Clients over LET-A Network

Video communication over FSMC enables us to study and explore the video transmission behaviour over LET-A as FSMC is such acceptable distance factor channel. So, we are introducing our collected results to reveal video transmission attitude over LET-A, which is our important gate for future work in this direction.

When using FSMC over Video application, this makes our application more realistic and functional, and then obtaining probability from FSMC is used over our work as a realistic probability where applied over our video transmission.

1.17 Thesis Organisation

The outline of the thesis indicating the relationships between different chapters is briefed below:

The introduction gives the problems this thesis tackles with our contributions. Moreover, it gives the essential knowledge which is needed to understand our work.

More detailed knowledge is explained in the background, allowing the reader to understand the techniques used in this research, before going for further specific technique issues.

After being sure that the reader understood our aims and contributions with the essential knowledge, we showed our first contribution in the third chapter. The third chapter shows two system designs, AF and DF and how applying NC over these two systems over the physical layer improves the bandwidth significantly, though we pay a little penalty in terms of BER loss to increase the reliability of transmission, Shannon limit approaching FEC Turbo code is proposed, which is Partial Unit Turbo Code. This turbo code is used as our recommended error-correction technique, which is applied in the physical layer by adding redundant bits to recover the information bits. Full details of PUMTC will be shown in Section 1.7.3. After showing how NC helps in bandwidth and data rate, we applied it over WSN as an application could harvest the benefits we obtained via applying NC.

Following to applying the NC over the physical layer, we extended NC applications to be over the packet level in the application layer, instead of bit level in the physical layer. This has been shown in the fourth chapter. NC is regarded as another error-control method in this chapter, which is mainly used on the packet level and operated across different packets. After performing the FEC over the physical layer, packets are regarded as either received with error free or discarded in the higher layers. Based on this, we assumed the erasure packet channel at the higher layers. Two different scenarios have been investigated in this chapter, NC cooperation among N users in the first data gathering WSN scenario where each user broadcasts its packet in the first stage, and then applies NC before relaying neighbour packets in the following stage. The second scenario was for downlink LTE-A with Femto relays, where the Femto relays receive those packets from the Pico relay, and then applying NC cooperative technique before forwarding them to the user.

The fifth chapter is aimed at FSMC erasure channel with time varying BER. This chapter introduced several type of NC with cooperative such as random and deterministic combinations to be applied over WSN as a proposed application. Moreover, the fifth chapter also models, via FSMC, packet losses for video distribution over LTE-A.

The sixth chapter focuses on the work conclusion and future work.

2. Background of Cooperative Network Coding Wireless Networks

2.1 Introduction

This chapter provides background and describes related research efforts in NC and Cooperation systems over wireless networks. Section 2.2 shows how a wireless sensor node is applied in real applications to illustrate the method where nodes are connected. After that, we start supporting this research with the background for the technologies used in this research, starting with FEC channel codes for error correction at the physical layer in Section 2.3 and subsections therein, such as linear block codes, convolutional code, Turbo code and then PUMTC. In Section 2.4, provides more details about NC such as random NC, deterministic NC, applications, types and benefits. Section 2.5 gives a background for the channels used in this research for both physical layer where communications suffering primarily from AWGN and multiplicative fading noise, and upper layer where communications suffering from packet losses or erasures.

More background is provided for LTE-A as it is the application we extended our FSMC over in Section 2.6. Followed by Section 2.7 which shows the channel modeling for video distribution over LTE-A. Finally, Section 2.8 summarises this chapter and highlights the research late chapters.

2.2 Wireless Sensor Nodes

The following example shown in Figure 2.1 illustrates an application which WSN could be used for, in either WAN or LAN as an example for wireless communications.

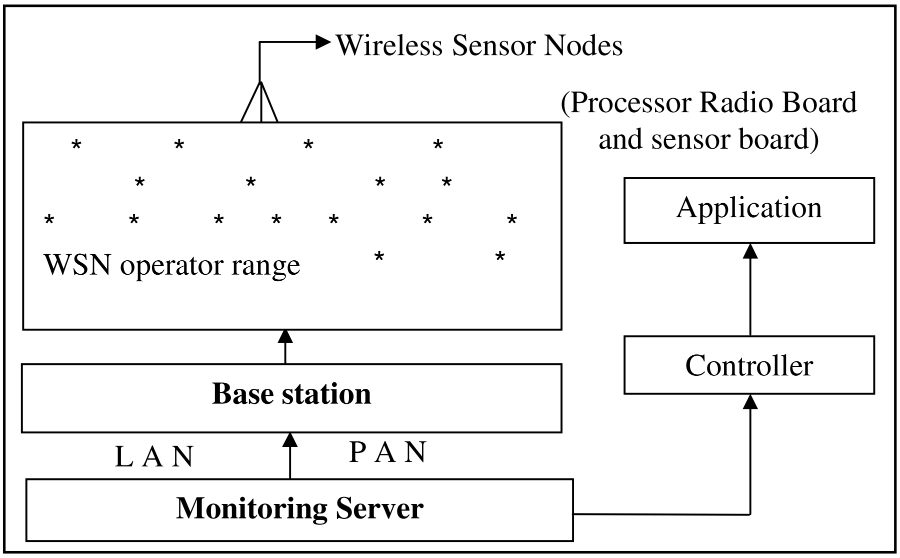


Figure 2.1: WSN Local Area Network or Personal Area Networking.

This example is for LAN WSN or PAN, for wider Network like PAN, we need routers to connect the sensor nodes with the base station.

The monitoring server could be data storage, display, analysis, decision support, or any application to monitor. WSN are all connected to the BS, and each node has usually a processor radio and sensor board.

So, we obtain the results at the monitoring server and then send the appropriate order to the controller to perform the application.

WSN in this application could be Internet client, light, microphone, acceleration EST. The base station could be programming and PC communication that changes the information from and to the sensor node. And finally, the WSN operator range is the place where WSN operates, i.e. where sensor nodes are fit, which could be an office, a building, hospital, etc.

For more information about WSN, like hardware, software, programming languages, conferences see Reference [65].

2.3 Forward Error Correction

Following to the FEC introduced in Section 1.5, more background is provided for linear block code and PUMTC used in our physical layer work.

The FEC used in the research is PUMTC, where PUM occupies an intermediate position between block and convolutional codes, as it will be shown in Section 2.3.1 and Section 1.7.3 restrictively. So, it is important to give good background for both block and convolutional codes below.

2.3.1 Linear Block Codes

The theory of linear block codes is well covered in many textbooks, such as [66], [67], [68], [69], and [70]. Following, we are introducing this technology showing the relative information needed in this research.

Linear block codes are the most widely used of block codes. This is justified by the simplicity in term of implementation. Linear block codes encoders transform a block of k messages digits into a longer block of n codeword digits. Therefore, these codes are characterized by (n,k). The added digits are called parity check digits p, which are used to mitigate the channel noise that changes sufficiently few of these n transmitted channel digits, i. e., the p check digits may provide the receiver with sufficient information to enable detecting and/or correcting the channel errors. When the codeword consists of 2 elements (0 and 1), the code is binary comprising binary digits.

The linear block codes have the following specifications:

- The combination of any set of code words results to a code word. Hence, linear block codes always contain the all-zero vector.
- The lowest-weight codeword is determined by the minimum distance of a linear block code. i. e., they are equal.
- The undetectable error patterns for a linear block code are independent of the code word transmitted and always consist of a set of all non-zero code words.

The two characters k and n determine the linear code generator matrix $\boldsymbol{G}$ that has $k \times n$ dimensions.

2.3.1.1 Linear Block Code Encoder

The codeword $\boldsymbol{c}$ at time t for the information symbols $\boldsymbol{u}$ is obtained from $\boldsymbol{G}$ as shown in

$$c^t = u^t .G , \tag{2.1}$$

where $\boldsymbol{G}$ is determined by (2.2)

$$G = [I_k \mid R], \tag{2.2}$$

where the identity matrix $\boldsymbol{I}$ has dimensions $k \times k$; and $\boldsymbol{R}$ generates the parity bits and has dimensions $k \times (n\text{-}k)$.

Moreover, (2.2) shows that $\boldsymbol{G}$ is systematic, i.e., all the input bits form a part of the output.

In general, each of the n digits can take one of q possible values. Although there are q^n different sequences of length n, only q^k of these sequences are codewords, because the r check digits within any codeword are completely determined by the k message digits. The set of these q^k codewords of length n is called the code. Linear codes are special family of codes that have the characteristic that the linear combination of any codewords is also a codeword.

2.3.1.2 Linear Block Code Decoder

To decode the codeword, the syndrome vector is usually calculated. To do so, the parity check matrix $\boldsymbol{H}$ of dimensions $(n\text{-}k) \times n$ is determined by (2.3):

$$H = [R^T \mid I_{(n-k)}], \tag{2.3}$$

where $\boldsymbol{R}^T$ is the transpose of $\boldsymbol{R}$.

Equations (2.2) and (2.3) show that there is a relationship between $\boldsymbol{G}$ and $\boldsymbol{H}$, which is given by equation (2.4).

$$H.G^T = G.H^T = 0, \tag{2.4}$$

where $\boldsymbol{H}^T$ is the transpose of $\boldsymbol{H}$.

When multiplying $\boldsymbol{H}^T$ with a valid codeword vector, a zero vector 'syndrome' is obtained.

The syndrome s is calculated as in (2.5)

$$s = r.H^T = (c + e).H^T = c.H^T + e.H^T, \tag{2.5}$$

where c is the transmitted codeword, r is the received codeword and e is the error vector. Equation (2.5) shows clearly that the actual value of the syndrome vector is only dependent on the error vector. Accordingly, the syndrome vector is used as an error detection factor by the decoder. Moreover, the syndrome's value would indicate the validity of the received codeword.

2.3.1.3 Linear Block Code Error Detection and Correction

When the syndrome's value is not zero, this means that there is an error faced the transmitted codeword. This error vector *e* can be calculated by the syndrome value, as there is a unique syndrome exists for every correctable error pattern. This technique is called syndrome decoding. The capability to control the error of a code is determined by its Hamming distance, where the Hamming distance is the minimum distance of a block code between all distinct pairs of code words. A code with minimum distance d_{min} can detect all error patterns of weight less than or equal to (d_{min}-1) but can only correct $\lfloor (d_{\min} - 1) / 2 \rfloor$ bit errors in a codeword.

Based on above, block coding can be introduced on as: (n, k, d), where $n=k+p$ and d is the hamming distance.

2.3.2 Turbo Codes

Turbo codes are class of codes designed for iterative decoding, which enables them to achieve near Shannon-limit performance [25] (Berrou et al., 1993). In fact, they are one of the closest to approaching the Shannon limit among the all practical error correction methods [44].

The main advantage of using Turbo codes is, the ability to increase data rate without increasing the power of a transmission, so, they can be used to decrease the amount of power used to transmit at a certain data rate.

These advantages in turbo codes cause the relatively high decoding complexity, moreover, it relatively increases the time needed to receive the sent packets (latency) [44].

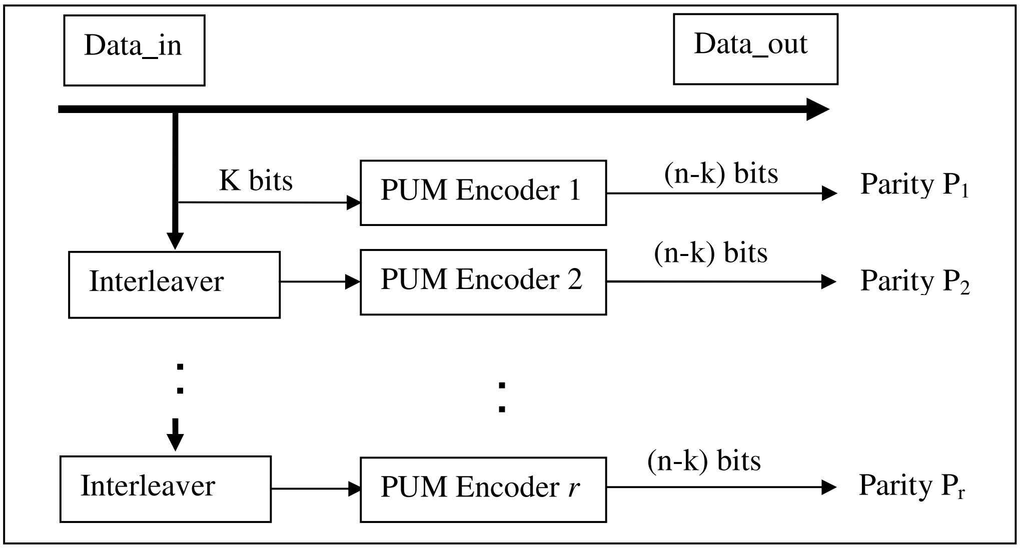


Figure 2.2: Block diagram of a PUM Turbo Code.

The interleaving provides r sets of parity bits which are uncorrelated and can be reduced by puncturing according to a puncturing matrix, e.g. if r=2, P_1 could be transmitted at time t_0 and not P_2, and then at time t_1, P_2 be transmitted and not P_1, and so on.

Moreover, the interleaving provides protection against burst error; indeed, spreading big number of faulty bits over different packets can make them all correctable in the new located packets.

2.3.3 Turbo Decoder

Figure 2.3 shows a typical turbo decoder with two decoders (r=2) For decoding accuracy, DEC1 decoder and DEC2 decoder must be able to accept and to deliver soft values, which called for the next decoding stage.

The reliability of the decoded bits depends on the adapted punctured coding, so some information bits are decoded without parity bits and others with the parity bits, which makes the frames to be not uniform [71]. The updated extrinsic information determines the performance of turbo decoding to correct the errors, so, the improvement of the extrinsic information by the iterative decoding is important to evaluate the reliability of this code [71]; in fact it is the key to high performance [39].

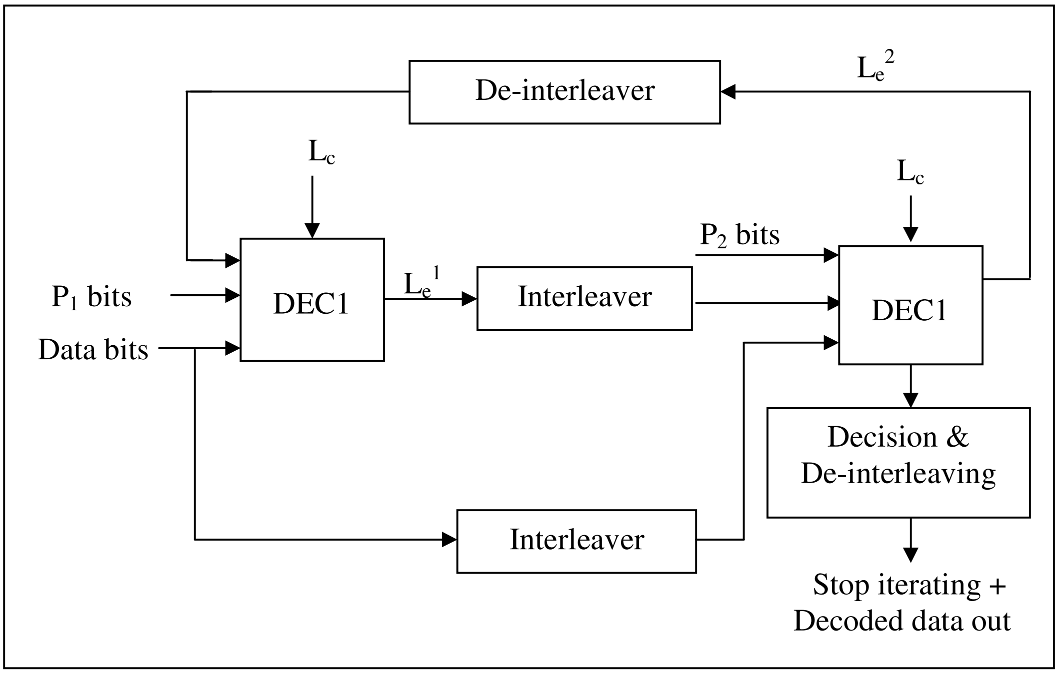


Figure 2.3: Turbo Decoder.

2.3.4 Turbo Codes Based on PUM

As mentioned in the previous section 2.3.4, PUM codes are dignified as a low-complexity alternative to the convolutional code which is one of the Shannon Limit Approaching codes. So, building turbo codes on PUM codes reduces the decoding complexity by reducing the number of the states in the trellis [39] and [41].

Accordingly, turbo codes with component PUM codes are considered as a low-complexity alternative to the classical turbo codes based on RSC codes [39].

Using PUM codes enables turbo codes to gain all the advantages mentioned in PUM, such as having fewer numbers of states than RSC codes in their trellis for the same number of encoder inputs, and the excellent distance properties [40].

So, using turbo code based on PUM ensures ‘that the performance of the PUM turbo codes will be maintained, if not improved, relative to the classical RSC turbo codes’ page 4, [39].

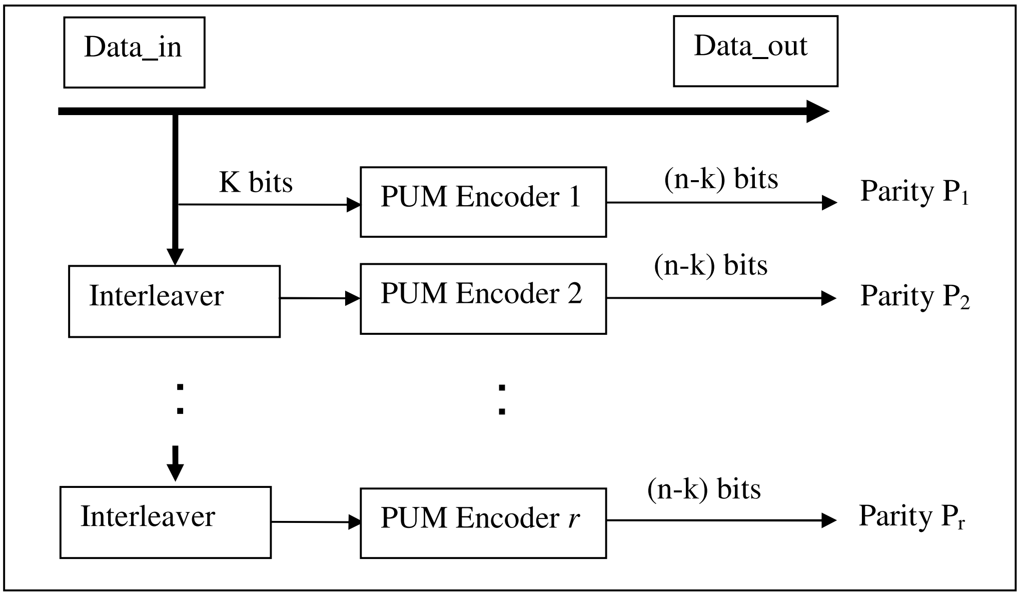


Figure 2.4: Block diagram of a PUM Turbo Code.

2.3.4.1 Turbo Encoder Based on PUM:

The structure of a PUM turbo code with *r* component (*n,k*) PUM encoders, which may or may not be identical, is shown in

Figure 2.4 [39]. All *r* component encoders use the same inputs, but in different sequences as a result of the interleaving.

Figure 2.4 shows a PUM turbo encoder with *r* component (*n,k*) PUM encoders.

2.3.4.1 Turbo Decoder Based on PUM:

Figure 2.3 shows a typical turbo decoder with two decoders (*r*=2), Which is similar to convolutional Turbo decoder shown, both PUM1 decoder and PUM2 decoder must be able to accept and to deliver soft values for the next decoding stage, to maintain the desirable accuracy.

The Turbo decoding reliability for the PUM depends on the adapted punctured coding as well. In fact, the rest of the decoding processing is similar to the convolutional Turbo decoding, apart from the fact that PUMTC is less complicated as a results for the less complicated encoding processing as was shown in Section 1.7.2) [39].

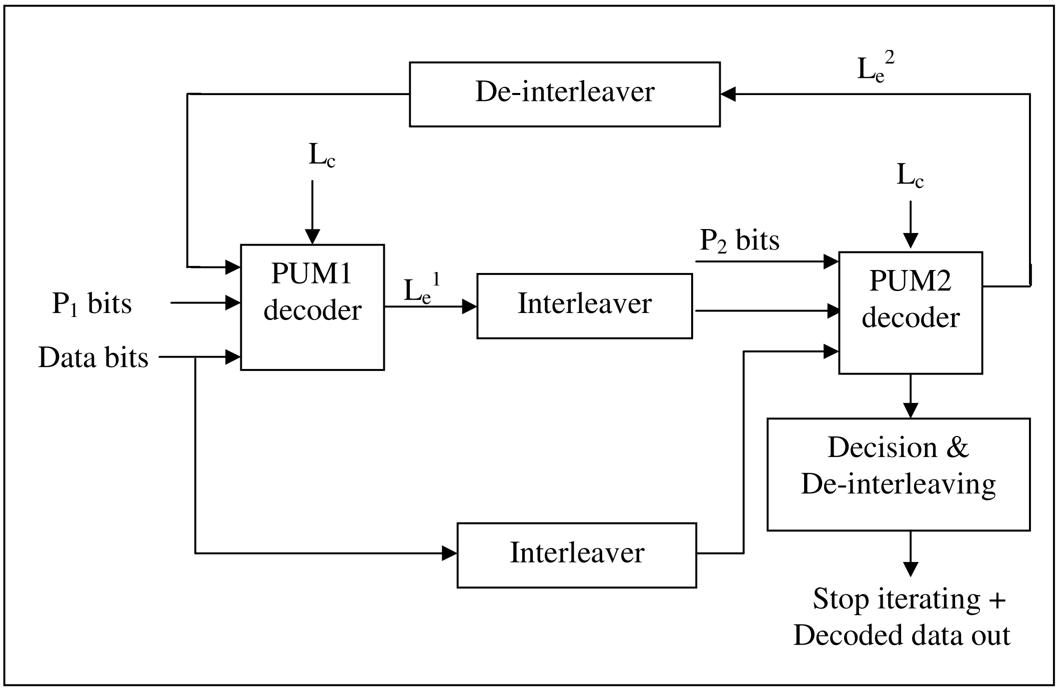


Figure 2.5: Turbo Decoder.

2.4 Network Coding

In this section, we describe random and deterministic NC.

2.4.1 Random Linear Network Coding

In this NC, the transmitters just tend to linearly combine a random number of packets, which is known as 'random linear mappings' [57].

In such case, in each incoming transmission, only the overall linear combination of source processes is needed to be known at the receivers. Each transmitted packet or block includes this information as a vector of coefficients corresponding to each of the source processes. At each coding node, the coefficient vector is updated as to the information signals by applying the same linear mappings. The transmitted overhead of these coefficients decreases with increasing length of blocks over which the codes and network remain constant. For example, in the case of a fixed network and code, the sources need to send the coefficients once at the start of operation through the network [57].

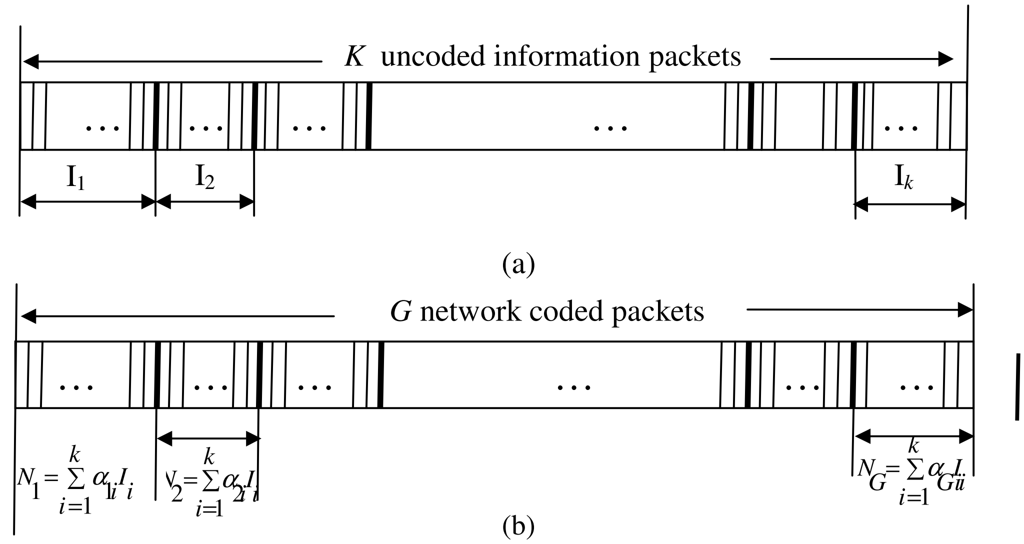


Figure 2.6: (a) *k* uncoded information packets; (b) *G* Random Network coded packets for *k*.

Figure 2.6 shows the way linear NC is applied over *k* uncoded information packets when random NC is applied over the *k* uncoded information packets with I_1, I_2, …, I_k vectors to generate *G* network coded packets denoted N_1, N_2,…, N_k as shown in (2.6).

$$N_j = \sum_{i=1}^{k} \alpha_{ji} I_i, \tag{2.6}$$

were α_{ji} ($i \in [1,2,...,k]$, $j \in [1,2,...,G]$) are scalars chosen uniformly from a Galios field of size q, denoted as GF_q.

The ration of the number of uncoded information packets k to the number of transmitted network coded packets G is denoted by NC rate R_n which is simply k/G.

2.4.2 Deterministic Network Coding

Deterministic NC, as reflected from the name, means combining packets linearly in deterministic way.

In such case, applying NC over the transfer matrix is reduced to choose just the appropriate elements (entries) rather than combining all the entire entries. The main requirement is that each sink can decode the entire source's information, i.e., it is essentially a requirement that the sink is able to invert its own transfer matrix, which means having full rank transfer matrix.

Based on above, we can confirm that the way the entries are selected to be combined must maintain the condition that the transfer matrices hold full rank at each sink simultaneously.

To solve this problem, we have proposed deterministic combinations where the transfer matrix is rank M at all sinks, where M is the number of the elements in the matrix, i.e., the transfer matrix has maximum rank. Moreover, we have proposed several transfer matrices give the less rank than M but it avoids the repetition when more than one combination is available, hence avoiding the redundancy in the transmission, when more than one relay to perform NC over the same elements or when more than one transmission stages is performed as it will be shown in chapter three.

In [72], a new deterministic algorithm to construct network codes for multicast problems have been proposed with an algorithm easily generalized to several variants of multicast problems. In our research we have proposed several deterministic combinations, such as, fully deterministic, odd-even combination, next neighbour only, etc.

These deterministic combinations showed that the deterministic way of combinations has several benefits, such as simplicity, and improving Packet Error Rate (PER), as it will be shown in the erasure channel results part.

Moreover, the deterministic combination requires a shorter headerfile than the random combination does because the sink needs to know the combination source than to know what is combined as in the random combination.

Gaussian Elimination is used to decode the required packets as it is always enough to have a received matrix with rank M to decode the M unknown packets, and our mail is to have this rank M in the fewest number of transmissions and maintaining it even in the case of lost some packets through the transmission.

2.5 Channels

In this part we expand on channel modelling, in the physical layer and upper layers, i.e., which model is for PHY layer and how it is represented to be used for MAC layer.

2.5.1 Additive White Gaussian noise

White Gaussian noise is a random process, each sample of which is a zero-mean Gaussian random variable with a level of $N_0/2$ Watts per Hertz flat power spectral density over the entire frequency range $-\infty \leq f \leq \infty$.

When transmitting a symbol x through A channel with AWGN, y symbol is received which is related to x by: $y = x + n_G$, where n_G is a zero-mean Gaussian random variable with variance σ_n^2 and the input x can have any one of L discrete values in a symbol alphabet size of $\log_2 L$.

2.5.2 Fading Channel

The motion between the transmitter and the receiver results to time variations channels, which leads to broadening of the signal spectrum. The Doppler spread f_D is defined by the frequency range where the power spectrum is non-zero. Equation (2.7) shows that f_D is proportional to the speed of the mobile unit and the carrier frequency:

$$f_D = \frac{v \times f_c}{c}, \tag{2.7}$$

where v is the speed of the mobile in ms^{-1}, f_c is the carrier frequency and c is the speed of light, i.e. $3 \cdot 10^8$ ms^{-1}.

Equation (2.8) summarizes the Rayleigh fading channel behaviour:

$$y = A.x + nG, \tag{2.8}$$

where A is the fading amplitude and the other parameters are as defined in Section 2.5.2. The Rayleigh channel is implemented using Jakes algorithm shown in [73].

2.5.3 Free Space Propagation Model

As WSN transmitter and receiver have a clear, unobstructed line-of-sight path between them, the radio wave propagation for WSN is described as free space propagation model. In this model, the received power is directly related to the transmission distance, i.e., the distance between the transmitter and receiver. This relation can be shaped by the power law function which states that the received power $P_r(d)$ decays as a function with distance as shown in the following equation:

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L}, \tag{2.9}$$

where P_t is the transmitted power, G_t is the transmitter antenna gain, G_r is the receiver antenna gain, λ is the wavelength, d is the transmission distance and L is a system loss factor not related to propagation ($L \geq 1{:}0$).

The losses are justified by the line attenuation, filter losses and antenna losses. The antenna gain depends on its effective aperture A_e which is related to the antenna physical size, so:

$$G = \frac{4\pi A_e}{\lambda^2}. \tag{2.10}$$

And λ depends on the carrier frequency f and the speed of light c by (2.11):

$$\lambda = \frac{c}{f} \tag{2.11}$$

Based on above, signal strength decays proportionally to the inverse of square of the transmission distance.

2.5.4 Finite-State Markov Chain Fading Channel

Markov chain is a discrete random process which has a discrete finite number of state-space. As Markov chain is a discrete random process, the system is in a certain state at each step, with the state changing randomly between steps. In wireless fading channels, the steps are considered as time, and called state-space, which are just integer numbers, and the random process is a mapping of these states. Moreover, the Markov current state depends only on the current state of the system, and not additionally on the state of the system at previous steps, so, it is generally impossible to predict the exact state of the system in the future. In wireless fading channels, the future state is either the current state or the adjacent. Change from one state to the next depends on the transition probability. The changes of system states are called transitions, the probabilities associated with various state-changes are called transition probabilities and the space-state BER reflects the channel quality at the transmission time. The set of all states and transition probabilities completely characterizes a Markov chain as shown in Figure 2.7.

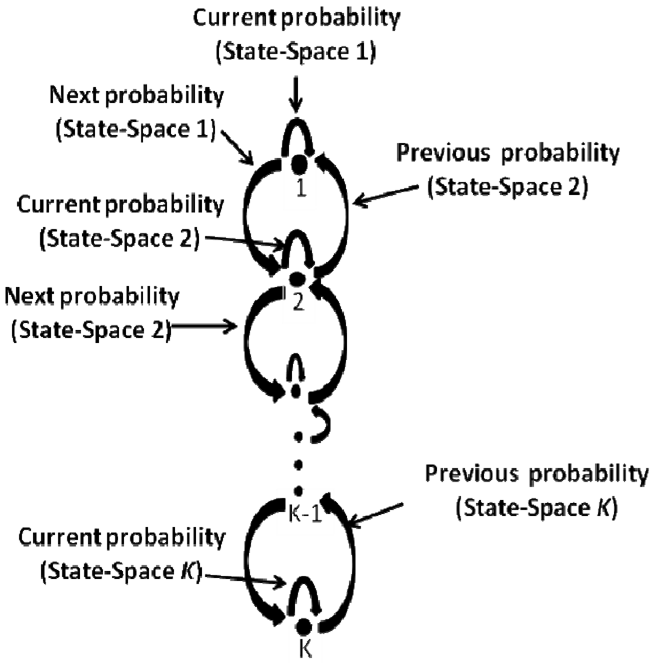


Figure 2.7: Markov chain for *K* state-spaces.

By convention, we assume all possible states and transitions have been included in the definition of the processes, so there is always a next-state and the process goes on forever.

2.6 Network Coding over LTE-Advanced

Instead of just transmitting the data packet separately, NC can directly manipulate the data packets whilst they are at a node in the network. This can be performed at the bit level in the physical layer or at the packet level (XOR-ing packets) in upper layers, this means that the data packets received at one node from different nodes can be combined together to be sent down the one channel, resulting to significant increase in the throughput.

Decreasing the number of downlink transmission in the cluster is an important advantage, because the cluster by nature, experiences a high amount of traffic which is vital to progression within this technology [13].

The most important part of this technique is the way we perform NC, i.e., the proposed combination in the intermediate node in the network must be performed in a way that allows the receiving node to be able to separate the packets properly and sequence them again. To enable the receiver node to do so, information data must be added to the network coded packets which called headerfile.

When NC works in a way to enhance the cooperative between nodes, we end up with a very reliable and power efficient system. Indeed, NC is a power effect technique although it takes extra power to perform the NC. This is because the power save outweighs the power usage [74]. Moreover, the range of communication can be significantly improved when cooperative NC is applied. These advantages and others, is the reason behind directing our research to CoMP and NC in 4G.

In our proposed NC Cooperative system, we showed how NC helps node to cooperate together through a relay though they are all out of range, and the significant improvement in the throughput for 4G Femto cells.

2.6.1 Real World Application

Such modern technologies are usually driven and funded by large commercial companies, mainly to advance their products, such as Samsung and Apple. The proposed technologies over 4G, so far have been tested experimentally using hardware devices such as computers and mobile phones. However, the real applications for these technologies are placed in common communication sectors, such as offices, homes,…,etc. Accordingly, it is helpful to extend the experimental test to cover the real world circumstances, to understand the way these techniques are applied.

Regarding CoMP, it is widely expected to be used in most future communication devices, as it is essential to build a fully heterogeneous network, which is the aim for the deployment process of 4G systems. Indeed, 4G heterogeneous network includes any devices connected to the system, such as mobiles, computers, wireless printers, computers and even modern cars can enhance their mobile signals when moving between cell sites through cooperation.

Moreover, the 4G heterogeneous network can consist of relay nodes, which are a step down from base stations and use the same spectrum as the backhaul system, but use similar power to smaller nodes. Other deployed nodes can be classified as:

- Macro cells – These are the conventional base station nodes, they use a dedicated backhaul and will usually allow public access. (Typical Transmit Power ~43dBm; Antenna Gain ~12-15dBi.)

- Pico cells – A lower power base station, also with dedicated backhaul connection and open to public access. (Typical Transmit Power ~23dBm; Antenna Gain – 0 5dBi.)

- Femto cells – Consumer deployable base stations that use the commercial broadband connection as backhaul. Possibly with a restricted access policy. (Typical Transmit Power <23dBm) [75].

- Relay cells – It is a base stations using the same spectrum as backhaul and access. Similar power as Pico's. Relay to Femtos provide coverage extension and capacity with little to no incremental backhaul expense.

Figure 2.8 [75] shows a full example for heterogeneous network deployments, and shows the types of low power bases stations deployed, which is one of the important defining features of LTE-A networks.

We can notice that the deployed BSs improve the Macro cell though out and improve coverage and capacity. Moreover, this approach increases the spectral efficiency per unit area.

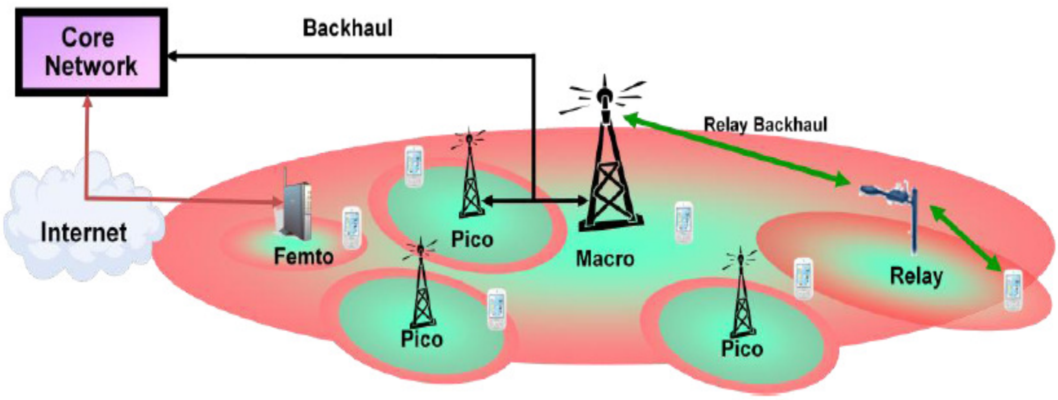


Figure 2.8: Heterogeneous Network utilizing mix of Macro, Pico, Femto and relay base stations.

The Femto cells are typically deployed in the home or office, and utilize the Digital Subscriber Line (DSL) or cable broadband as a wired backhaul into the core network.
They are very low power in some cases downs to ~10mW, with a coverage radius in tens of meters and limited capacity. Unlike Femto cells, Pico cells have a larger coverage area and dedicated backhaul, so they can be employed in places like urban hotspots to supplement the capacity of the Macro cell base station.
A potential problem is unwanted signals received from other base stations which will require advanced interference management. One strategy is the combining of information from multiple sources and forwarding it on instead of treating it as interference.
Regarding HARQ, it is being used in the 3G equivalent of LTE, WiMax and Universal Mobile Telecommunications System (UMTS) already and will continue to be used as the technologies progress to their 4G deployment. Moreover, the latest IEEE 802.11 standards released for wireless systems – 802.11n – has been amended to include details of Multiple Input Multiple output (MIMO), so HARQ is becoming more relevant and widely used as time goes on.

Out of complexity and implementation difficulties, NC practically, will not be as widely used as CoMP and HARQ in 4G devices overall in nowadays real work applications. However, NC can be implemented in 4G under certain circumstances, such as when retransmissions are particularly costly to system performance.
In our proposed research, we focus on NC as an alternative way to decrease the retransmission. In addition, NC results to significant decrease in the transmission traffic which is a key factor in clusters' communications, as traffic is a vital to aggregation process.
Finally, our proposed research shows that NC can manipulate data packets in such simple and deterministic way, resulting to a practical and power efficient system with significant throughput improvement.
Applying NC in downlink transmission results to a very well-cooperated system, and this is one of our proposed research areas as too.

2.7 Channel Modelling for Video Distribution over LTE-A

It is no secret that video transmission over wireless networks faces many challenges, such as packet losses, bandwidth limitations on the time-varying channel; which is our research channel. To mitigate the problem of packet losses to improve the quality of video reconstruction; error correction and error resilience mechanisms are widely implemented. One such scheme is Multiple Description Coding (MDC) [76], which exploits the path diversity and then improves the reconstruction quality.

The procedures start by encoding the source data in multiple descriptions, and then send the encoded video data streams into the receiver over independent paths. In the other side, the receiver reconstructs the received encoded data, at an acceptable quality for the received subset descriptions, taking into consideration that, the more received descriptions, the better the reconstruction quality. This enables the system to have video rendering at the receiver despite a complete path failure.

Random Linear Codes (RLC) [77] recently became a popular class of rateless codes, which has been applied over the source message to produce the encoded symbols in term of random linear combinations. As shown in Section 0, RLC is applied over the source message to obtain linear combinations multiplied with randomly selected coefficients from a given finite field. In the packet level Application-Layer Forward Error Correction (AL-FEC) solution; RLC is regarded as a simple to implement with near-optimal erasure codes performance. Moreover, the decoding Complexity via using Gaussian Elimination is acceptable in the case of short length source messages [78] and references therein. These advantages enhance RLC to be a competitive candidate for the universal FEC/network coding solution even in the most recent video transmissions over wireless communication systems, such as LTE-A, WiMAX, and Digital Video Broadcast- Next Generation Handheld (DVB-NGH).

The multiple descriptions are normally created by duplicating the important data over both descriptions resulting to Unequal Error Protection (UEP) to ensure a better reconstruction quality. In [79], redundant slices are used for creating multiple descriptions, where in [80], the base layer and the enhancement layer packets are sent over different paths and Selective ARQ is used to notify base layer loses when the re-transmission is needed. However, it is difficult to cope with long error bursts for any scheme which is non-adaptive to the channel. To help in solving this problem of ARQ, we have proposed applying NC over the LTE-A network, precisely, over the Femto relays. Indeed, when we apply our proposed deterministic NC over the Femto relays rather than just forwarding the received packets without applying NC; the probability for the need of ARQ has been decreased significantly, as the sinks (users) gain the ability to retrieve the unreceived packets rather than just asking for ARQ when not receiving any packets.

2.8 Summary

This chapter provides background and describes related research efforts in NC and Cooperation system over wireless communication. Main concepts have been explained such as WSN Section 2.2, technologies used in this research in Section 2.3, NC types, applications and benefits in Section 2.4, channels used in this research in Section 2.5 LTE-A in Section 2.6, channel modeling for video distribution over LTE-A in Section 2.7 and the summary in Section 2.8.

Next Chapter introduces our work over the physical layer, using PUMTC as channel coding FEC, work show systems when NC is applied over AF and DF relay transmissions and compared with the benchmark systems when NC is not applied.

Next Chapter introduces our work over the physical layer, using PUMTC as channel coding FEC, work show systems when NC is applied over AF and DF relay transmissions and compared with the benchmark systems when NC is not applied.

3. Physical Layer Cooperative Network Coding

3.1 Introduction

In this chapter we propose two practical power- and bandwidth-efficient systems based on AF and DF schemes to address the problem of information exchange via a relay. The key idea is to channel encode each source's message by using a high-performance non-binary turbo code based on PUM codes to enhance the BER performance, then reduce the consumed energy and increase spectrum efficiency by using NC to combine individual nodes' messages. Two simple and low complexity physical layer NC schemes are proposed based on spatial and temporal combinations of received source messages at the relay. Simulation results under AWGN confirm that the proposed scheme achieves significant bandwidth savings and fewer transmissions over the benchmark system which does not resort to NC. We also propose a cooperative strategy that is useful when insufficient combined messages are not received at a node to recover the desired source messages.

Power-constrained WSN, with applications ranging from battlefield surveillance, medical care to environmental monitoring, are, in general, composed of many small sensor nodes with limited lifetime (i.e. battery power), hence, power-efficient protocols that reduce the node power-consumption by cutting down on communications which are a key requirement for practical WSN applications. This direction motivated research into power-efficient protocols with minimal communications overhead and relay-based approaches to extend the coverage area of the WSN via novel techniques such as NC (NC) [81], cooperative communications [82] and [83], and cooperative NC [84]. An efficient implementation of NC with low computational power is presented in [81]. In

[82], network cooperative communications has been investigated for QoS provisioning in resource-constrained WSN and proposes amulti-agent reinforcement learning based multi-hop mesh cooperative communication mechanism. Both NC and cooperative techniques are proposed in [84], analyzing the relay's location and resulting in increased coverage area. Moreover, cooperative diversity, where nodes relay each others' messages to achieve spatial diversity, by forming a virtual MIMO antennas between nodes in WSN has been investigated resulting in significant saving in transmit power such as in [82] and [82] and references therein. The NC approach is gaining popularity in WSN [6] as an extension to traditional routing techniques to allow nodes, termed encoding/intermediate nodes in contrast to traditional forwarding nodes, to mix the information content of received packets before forwarding them to destination nodes in the network. NC ingenuity comes not only from its classic throughput enhancement but also its significant energy saving reflected by the reduced number of transmissions required to deliver a packet compared to traditional routing.

In this part of research, we build on [82] where a full-duplex physical layer NC scheme is proposed for a three-node network comprising two sources which want to share their information via a relay. Results using pseudo-random and quasi-cyclic regular LDPC codes show that, instead of two separate transmissions from the relay, only one transmission is needed, which decreases power consumption and bandwidth and increases the communications range of the two sources. Indeed, if we suppose that x_1 and x_2 are two bit streams of equal length, and each transmission consumes the same amount of power, then the length of the combined packet from the relay is the same as the length of x_1 or (x_2), and the required transmitter power is halved if the two streams are separately transmitted.

In this work we propose a physical layer NC scheme combined AF DF cooperation strategies is implemented with a practical error control code, namely non-binary PUMTC [82], to exchange data by exploiting the broadcast nature of wireless radio links. BER and EXIT chart performance analysis [89] show that PUMTC outperforms the well-known turbo codes based on binary recursive convolutional codes. Moreover, PUMTC can achieve acceptable BER performance with smaller block sizes than LDPC codes, and is simple and robust enough for WSN.

Two practical system design schemes are proposed based on PUMTC, and compared to classical setups that do not exploit NC, assuming AWGN channels. The first system resembles AF relaying, where the relay does not perform decoding: it simply relays the received signals. In the second system, based on DF, the relay decodes received signals, before relaying on the reconstructions. The first system is more suitable for real-time communications since decoding at the relay is avoided.

In related work, channel coding and NC are combined for one-way communication with one intermediate relay node in [89]. More recently, two-way wireless communication was considered in [90], [91], and [92]. In DF scheme of [90], distributed turbo codes were used for protection: each node receives data from the relay and directly from the other node over two orthogonal channels; joint decoding is used for reconstruction for each node. The benefit of combining NC with convolutional codes via DF was shown in with another technique, denoise-and-forward, which improves AF, was developed in [92]. Physical Layer Network Coding (PLNC) schemes are shown in [93] to be suitable for multipath propagation applications with potential doubling of the network capacity of bi-directional communication between pairs of end users connected by a relay terminal in an AWGN channel. Similarly, [94] shows that the ergodic capacity of the cooperative relay networking scheme is slightly better in comparison with the Analogue NC scheme due to diversity combining gain in cooperative relaying. Practical and capacity approaching PLNC schemes over two-way relay channels, are proposed in [95] with a superimposed XOR PLNC scheme, tailored for asymmetric broadcast channels. Achievable rates are derived in [96] for the multiple-parallel relay channel using the max−flow−min−cut bound, DF, partial DF, Compress−and−Forward and Linear Relaying protocols showing that DF gives the highest capacity results using signal regeneration at the relays. El Gammal *et al.* in [97] establish upper and lower bounds on the capacity and minimum energy-per-bit for general and frequency-division AWGN relay channel models, correcting some previous theorems and introducing the best upper bound to the lower bound capacity theoretical limits for various systems.

3.2 Capacity of Proposed Systems

We consider a two-way communication scenario for exchanging messages among N source nodes via a relay. Each source node generates a message that needs to be delivered to all other nodes in the network. This scenario can emerge in wireless sensor and actuator networks or Internet of nodes where each intelligent source node must be aware of the measurements at all other nodes in order to act on them. To reduce power consumption, all communications take place via the relay. In the following we assume perfect synchronization among the nodes which can be achieved via GPS or synchronization pilot signals that can also be used for channel estimation.

Node i, $i = 1,2,...,N$, generates its message m_i, encodes it using an ideal Gaussian codebook and sends the resulting i.i.d. signal x_i with power P_i over a wireless channel (which, for simplicity, is modeled as an AWGN channel) to the relay. We assume that messages m_i are uniformly distributed binary sequences independent of the messages generated by other source nodes and of channel noise. The uplink channels, connecting the N source nodes to the relay are orthogonal. Thus, for $i = 1, 2,..., N,$ the N signals received at the relay are:

$$y_i = x_i + z_i^{UL}, \tag{3.1}$$

where z_i^{UL} is the uplink i.i.d. Gaussian noise of unit power independent of the source signals. The relay collects signals from all N source nodes, $y_1, \ldots, y_N$, and forwards by broadcasting to all nodes i, where y_j, $j \neq i$. To do that, the relay can resort to either AF or DF strategies.

In Section 3.2.1, for each of the two forwarding techniques, AF and DF, we give the limits for both systems: the proposed schemes based on NC and the corresponding benchmark systems that do not exploit NC.

3.2.1 Traditional Benchmark Schemes Based on AF and DF

In AF, the relay only amplifies the N signals it has received before forwarding to the N nodes. Then, the received signal at the i^{th} node is:

$$\hat{y}_i = A_{AFb} y_i + z^{DL} = A_{AFb}\left(x_i + z_i^{UL} \right) + z^{DL}, \tag{3.2}$$

where A_{AFb} is the amplification factor at the relay, and z^{DL} represents AWGN in the down-link (DL) channel. Note that the relay needs to broadcast *N* unique packets.

In DF, the relay decodes the received *N* signals, re-encodes, modulates, and amplifies them, and then forwards the *N* resulting signals. The signal received at the i^{th} source node is:

$$\hat{y}_i = A_{DFb}\, \hat{x}_i + z^{DL} , \tag{3.3}$$

where $\hat{x}_i$ is the re-encoded and modulated signal originating from source node *i*.

The above benchmark systems for AF and DF are illustrated in Figure 3.1 for *N=4* number of source nodes as an example. Figure 3.1 (a) shows Benchmark AF (AF_b) where the signal received by any Node *i* , $1 \le i \le 4$ is given by (3.2).

Thus, the overall capacity per node, in the AF_b, mode for any number N of nodes is given by (3.4), since any node will only receive one information bit per transmission.

$$C_{AFbi} = \frac{1}{2}\log\left(1 + \frac{A_{AFb}^2 P_i}{A_{AFb}^2 + 1} \right). \tag{3.4}$$

The DF Benchmark Decode-and-Forward (DF_b) system is shown in Figure 3.1 (b), where the relay encodes separately y_1 to y_4, reconstructing m_1' to m_4', which are re-encoded and modulated as $\hat{x}_1$ to $\hat{x}_4$, respectively and then amplified with gain A_{DFb} before broadcasting. Note that, the relay does not need to use the same codebook as the source nodes. The signals received by all nodes can be obtained from (3.3).

The capacity of the uplink channel between Node *i* and the relay can be derived from (3.1) to be $\frac{1}{2}\log\left(1 + P_i\right)$ and the capacity for the downlink channel derived from (3.3) is $\frac{1}{2}\log\left(1 + A_{DF_b}^2 P_i \right)$.

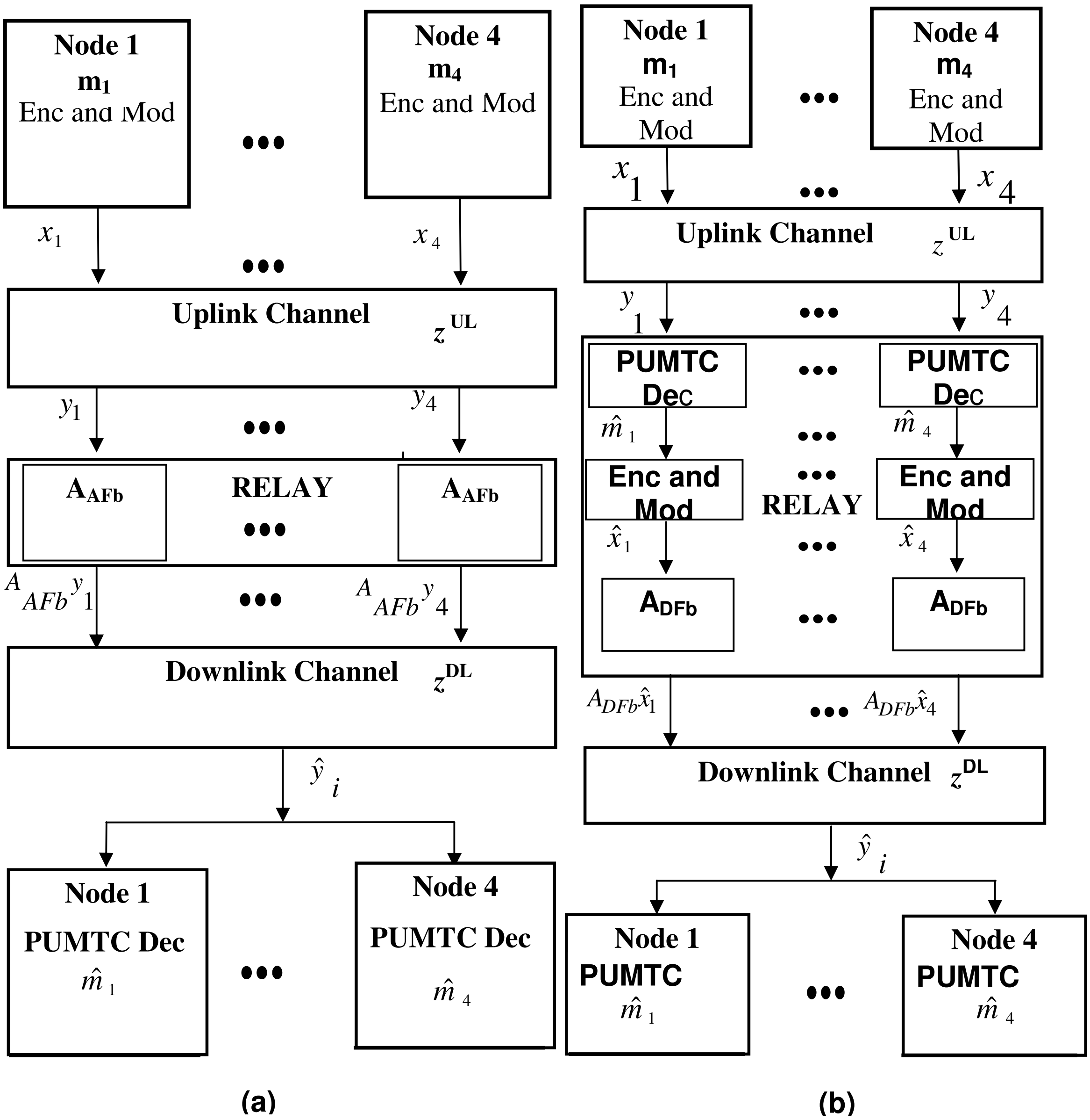


Figure 3.1: (a) Amplify-and-forward benchmark system, (b) Decode-and-forward benchmark system.

The overall capacity is the minimum of the capacities in the uplink and the downlink channel. Since $A_{DFb} \geq 1$, the overall capacity is dictated by the uplink:

$$C_{DFb} = \frac{1}{2}\log\left(1 + P_i\right) \tag{3.5}$$

3.2.2 Proposed AF and DF Schemes Based on Network Coding (Users in Range with the Relay)

Our proposed schemes show that applying NC deterministically before broadcasting combined y_i packets received at the relay can result in a gain in the data rate and a more reliable system in terms of cooperation among the nodes, fully exploiting the broadcast nature of the wireless channel. Traditionally, exchanging data between N nodes via a relay requires a total of $N(N\text{-}1)$ separate DL transmissions if no broadcast mode is available, or N transmissions by using broadcasting as is typical in WSN.

The relay "handles" multiple streams by using either time sharing or data mixing schemes (i.e. NC) [98]. The proposed system brings together the two schemes by first combining y_i from two sources received after the first UL transmission at the relay, and then broadcasting no more than $N\text{-}1$ combined packets in $N\text{-}1$ time slots. The combination at the relay is in the form: $y_1+y_2,\ y_2+y_3,\ \ldots,\ y_{(N-1)}+y_N$, taking into account that x_i is known at the i^{th} node and other x's can be recovered from received packets.

In the proposed AF scheme (AF_p), the combined packet received after AF_p broadcasting at the j^{th} time slot is:

$$\hat{y}_j = A_{AFp}\left(x_i + z_i^{UL} + x_{i+1} + z_{i+1}^{UL}\right) + z_j^{DL}, i = 1,\ldots,N-1, j = i, \tag{3.6}$$

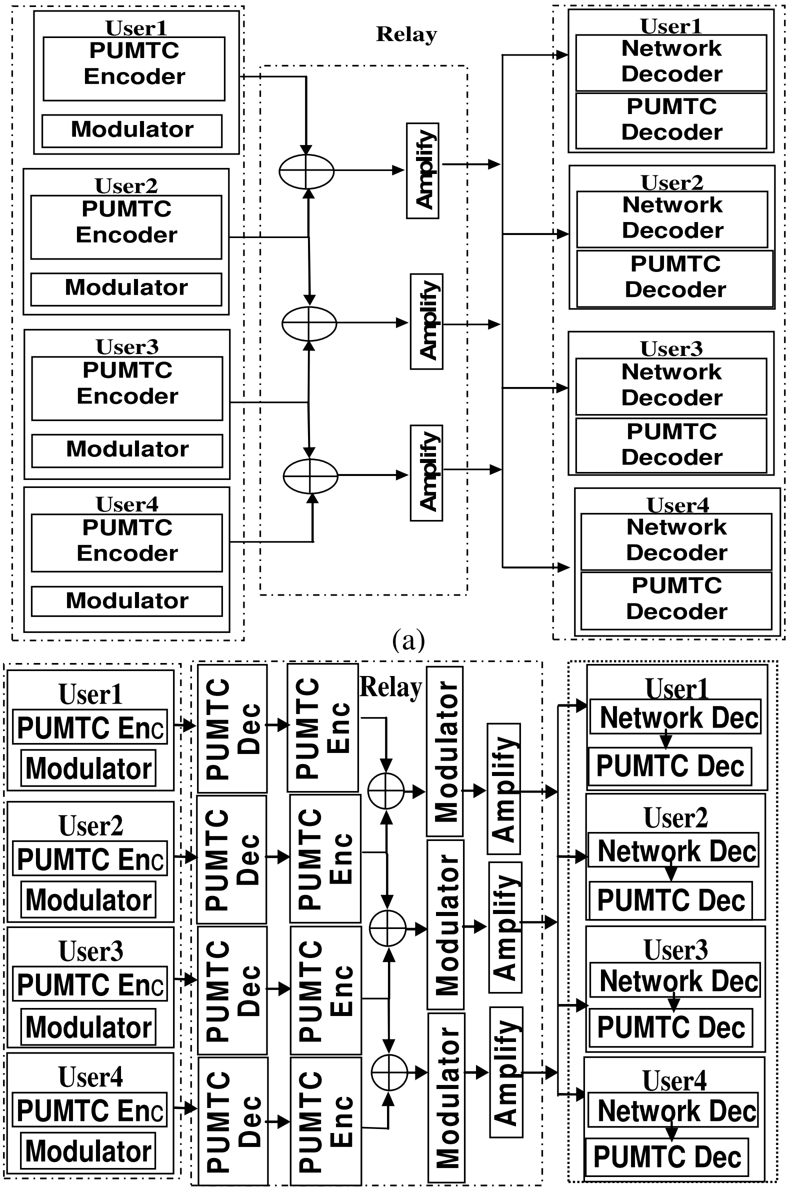
where z_i^{UL} and z_j^{DL} refer to AWGN during UL transmission from the i^{th} user and DL transmission at the j^{th} time slot, respectively. $A_{AFp} \geq 1$ is the gain assigned by the relay prior to forwarding the combination of N-1 noisy combined packets received from sources i and i+1. As shown in (3.6), the relay transmits the sum of the first two y_i's in the first time slots, and so forth, hence $j=i$. Each node must receive the same N-1 messages to recover all partners' messages. Moreover, (3.6) shows that the capacity per source node during the AF_P scheme for the proposed system is as (3.7), where the capacity per source node is the minimum of all transmissions. C_{AFpi} is identical for all i, as any source decodes only one message per received combined message comprising no more than two combined packets.

$$C_{AFpi} = \frac{1}{2}\log\left(1 + \frac{A_{AFb}^2 P_i}{2A_{AFb}^2 + 1}\right). \tag{3.7}$$

The proposed DF scheme (DF_p) adds a combination step to the benchmark DF_b between encoding and modulation. DF_p is summarized in (3.8), where $A_{DFp} \geq 1$ is the gain and the combination is a simple XOR operation. The node capacity for DF_p is equal to that of DF_b but overall with N-1 DL transmitted packets and a higher data rate.

$$\hat{y}_{(i,i+1)_{DFp}} = A_{DFp}\left(\hat{x}_i + \hat{x}_{(i+1)}\right) + z_j^{DL}, i = 1,\ldots, N-1; j = i. \tag{3.8}$$

Figure 3.2 (a) and (b) summarize the proposed AF_p and DF_p with N=4 as an example. Figure 3.2 shows that only three DL transmitted packets are needed to connect four nodes, compared to four packets without NC. In other words, the proposed AF_p and DF_p schemes reduce the number of DL transmissions by 25% for N =4, and, in general, (100/N)% for N nodes. For small N, this results in significant savings in transmission costs. It is important to note that the combination during NC simply sums noisy packets or decoded and modulated the received packets during AF_p and DF_p, respectively, with no concatenation and no extra header information requirement since combination is deterministic. While there is no change in capacity with the proposed DF compared to DF_b, the capacity of the proposed AF system is less than that of AF_p due to the accumulation of noise during combination at the relay.



(b)

Figure 3.2: Proposed PLNC schemes for 4 nodes, using: (a) AF_p (b) DF_p relaying strategies respectively.

3.2.3 Message Recovery for Proposed Schemes

Each node i wishes to recover the estimated $\hat{x}_k$ received by the relay during UL where k=1,2,…,N and $k \neq i$, using the received N-1 packets broadcast by the relay in the AF_p and DF_p systems as given by (3.6) and (3.8), respectively.

The message recovery process uses the fact that x_i is known by node i and reverse engineers the network encoding process by 'subtracting' the known message from the received noisy stream as expressed in (3.9) and (3.10) for AF_p and DF_p systems, respectively, where k=i+1.

$$x'_k = A_{AFp}\left(x_i + z_i^{UL} + x_{i+1} + z_{i+1}^{UL}\right) + z_j^{DL} - A_{AFp}x_i \tag{3.9}$$
$$= A_{AFp}x_{i+1} + A_{AFp}\left(z_i^{UL} + z_{i+1}^{DL}\right) + z_j^{DL},$$

$$x'_k = A_{DFp}\left(\hat{x}_i + \hat{x}_{i+1}\right) + z^{DL} - A_{DFp}x_i. \tag{3.10}$$

Recovery via AF_p will yield a noisier and less reliable $\hat{m}_k$ than DF_p. DF_p relies on a good channel code such as PUMTC to ensure that $\hat{x}_i$ is error-free, i.e., $\hat{x}_i = x_i$.

In traditional linear NC the encoded packets at the destination nodes are decoded using the Gaussian Elimination Algorithm (GEA), in which a set of linear equations that are formed of linearly independent encoding vectors $\{g_1, \dots g_N\}$ where $g_i \in \{0,1\}$ is chosen over Galois Field (GF) $\mathbf{F}_2$, and encoded packets are stored row by row in a decoding matrix. Initially, each row contains the original packet of the decoding node and the corresponding independent encoding vector, and GEA is used to solve the system of N linear equations. Similarly, the Gauss-Jordan Elimination Algorithm (GJEA), a variation of GEA, solves the linear equations by inserting zeros both above and below each non zero (pivot) element (e.g., ones) as it goes from the top row of the given matrix to the bottom.

In this scenario, we use a modified version of the GJEA where first, in each row of the decoding matrix there are two pivot elements representing the two combined encoded packets; then the decoding process starts from the row corresponding to the decoding node unlike the original GJEA that starts from the first top row. For example, to decode received packets at the third node, the pivot element representing the third packet is zeroed, since the third original packet is already known by the node; we then solve the pivot element representing the packet of the fourth node. Then similar to GJEA, zeroes are inserted both above and below for known packets and we solve for the remaining unknown packets. This modified algorithm saves computation resources compared to the classic GJEA because only N-1 computations are needed, as illustrated in Figure 3.3 (a). Node 1 aims to recover N-1 messages from all other nodes, given that x_1 is known at Node 1. First, Node 1 recovers x Combined packets or AF_p or DF_p, respectively then x'_2 is used to recover x'_3. This sequential process continues until all remaining unknown x'_k for k=4,…,N are recovered. The recovered x'_k are decoded via the PUMTC decoder to estimate the original messages $\hat{m}_k$. The operation flow of our decoding algorithm is shown in Figure 3.3 (b).

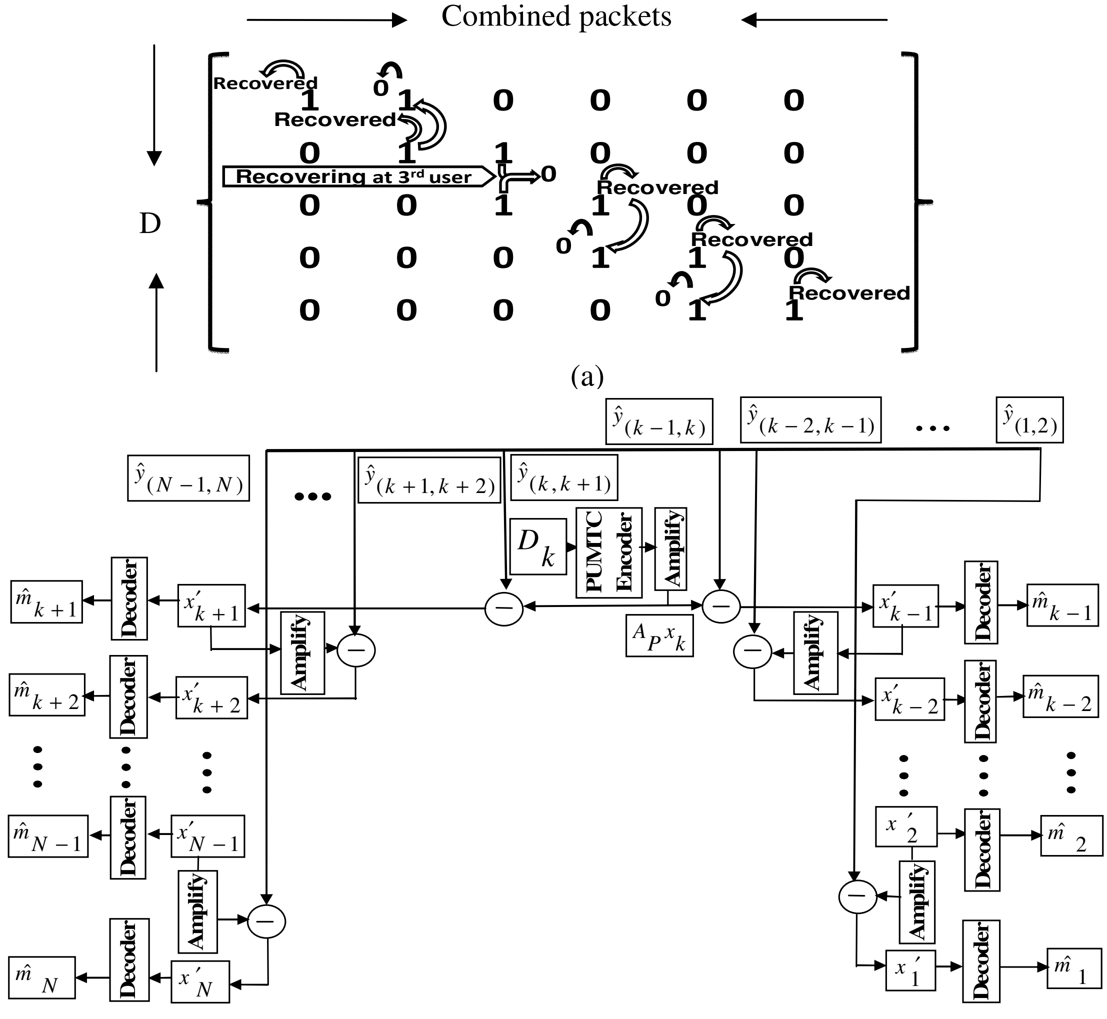


Figure 3.3: (a): Gauss-Jordan elimination for N=6 (with N-1=5 broadcast NC transmissions) at the 3rd node. (b): Network decoding processes.

Recovery steps of network coded messages are split as top and bottom elimination in Figure 3.3 (a), and left and right branches in Figure 3.3 (b). Starting from any node k, there are two directions to recover unknown x_i, starting with the known x_k and then determining the estimated received messages from the right branch (estimated packets from nodes labeled with indices less than k), and the estimated received messages from the left branch. Note that estimating x_i' for the left branch can be carried out in parallel with right branch estimations.

The processing steps at node k can be summarized by (3.11) and (3.12), for the left and right branches, respectively, where A_p refers to the gain A_{AFp} or A_{DFp}, depending on which scheme is used and i= 1,2, …,(N-k).

$$x'_{k+i} = \hat{y}_{k+i-1,k+i} - A_p x'_{k+i-1}, \tag{3.11}$$

where

$$x'_{k-i} = \hat{y}_{k-i,k-i+1} - A_p x'_{k-i+1}. \tag{3.12}$$

According to Figure 3.3 (a) and Figure 3.3 (b) and to (3.11) and (3.12), the more nodes in the system, the more recovery steps needed, which means potentially more error propagation.

The number of recovery steps at the receiving side can be reduced if the relay broadcasts additional packets C_T as shown in (3.13) and (3.14) for the i^{th} node for the DF_p example. The same principle stands for AF_p.

$$C_{T(k,k+i+1)} = A_{DFp}\left(\hat{x}_k + \hat{x}_{k+i+1}\right) \quad ,i = 1,2,\ldots,N-k\text{-}1, \tag{3.13}$$

$$C_{T(k,k-i-1)} = A_{DFp}\left(\hat{x}_k + \hat{x}_{k-i-1}\right) \quad ,i = 1,2,\ldots,N\text{-}k\text{-}2. \tag{3.14}$$

For example, for N=4, if Nodes 1 and 4 want to recover x_4 and x_1, respectively, previously, both x_2' and x_3' must be recovered first, resulting in error propagation and higher bit-error rate for both of $\hat{m}_1$ and $\hat{m}_4$, as shown in the simulation results section. Sending additional packets $C_{T(1,4)}$ removes the need to recover both x_2' and x_3' first.

In fact, these additional C_T transmissions ensure efficient ARQ when source nodes request missing packets at the relay, i.e. the relay can effectively combine requested packets by source nodes instead of just broadcasting them separately

Further cooperation at the relay is next discussed, showing how applying NC over the relay saves the requested number ARQ packets by the N users when some packets are not received by a node.

3.2.4 Cooperative Network Coding

In this section, we allow a one packet extra redundancy for the NC protocol proposed above. So, the relay broadcasts N combined packets instead of N -1 network-coded packets, in a cooperative manner to address the fact that some packets might not be received at any source. This extra packet still follows the adjacent combination principle used previously but in a circular fashion. We extend our proposed schemes by combining and broadcasting N packets as opposed to N-1, in a cooperative manner rather than the traditional selfish uncombined forwarding technique of the benchmark systems.

Each node receives N combined packets and aims to recover N-1 unknown messages from other N-1 nodes. Since packets are linearly combined, a node needs only N-1 combined packets out of the N broadcasted packets to recover the N-1 unknown x_i, i.e., one missing packet does not hinder recovery of all N packets, resulting to the fact that no ARQ request is needed. The packet recovery process is achieved by using the proposed modified GJEA and carried out in the same way as in Section 3.2.3.

If any source node is missing more than one packet, it can still recover the missing messages by sending ARQ requests to the relay. Therefore, each node requests the missing packets from the relay separately. The relay compiles all requests from all nodes in a histogram, and only broadcasts the missing combined packet with the highest demand. The nodes count the packets that they are still missing for recovery and send requests to the relay as before, which after compiling the histogram, broadcasts the packet with the highest demand in the next step. This process is carried out until N-1 unique packet combinations are received by each node. Cooperative broadcasting with NC as above reduces the number of retransmissions from the relay.

The following example shows how many ARQ requests from each source are required with the proposed scheme when two packets are not received by each node. In this example, N=10. Table 3.1 shows which packet combinations were not received. Only one packet is needed out of the two missing packets for any node to recover all other nodes' messages. The histogram of Figure 3.4 (a) shows how many of which packets are needed from the relay via ARQ from each step. At first, packet combinations y_3+y_4, y_5+y_6, and y_6+y_7 are most needed. Therefore, we randomly pick any one of these three; in our example, we retransmit y_5+y_6.

Table 3.1: ARQ requests to the relay when any two random packets are not received per source node.

<table>
<tr><th>Node</th><th>1</th><th>2</th><th>3</th><th>4</th><th>5</th><th>6</th><th>7</th><th>8</th><th>9</th><th>10</th></tr>
<tr><td>Not received</td><td>y_3+y_4, y_6+y_7</td><td>y_2+y_3, y_5+y_6</td><td>y_4+y_5, $y_{10}+y_1$</td><td>y_7+y_8, y_6+y_7</td><td>y_9+y_{10}, y_4+y_5</td><td>y_5+y_6, y_7+y_8</td><td>y_1+y_2, y_3+y_4</td><td>y_8+y_9, y_5+y_6</td><td>y_3+y_4, $y_{10}+y_1$</td><td>y_2+y_3, y_6+y_7</td></tr>
<tr><td colspan="11">Step 1 => Retransmit y_5+y_6</td></tr>
<tr><td>Not received</td><td>y_3+y, y_6+y_7</td><td></td><td>y_4+y_5, $y_{10}+y_1$</td><td>y_7+y_8, y_6+y_7</td><td>y_9+y_{10} y_4+y_5</td><td></td><td>y_1+y_2, y_3+y_4</td><td></td><td>y_3+y_4, $y_{10}+y_1$</td><td>y_2+y_3, y_6+y_7</td></tr>
<tr><td colspan="11">Step 2 => Retransmit y_6+y_7</td></tr>
<tr><td>Not received</td><td></td><td></td><td>y_4+y_5 $y_{10}+y_1$</td><td></td><td>y_9+y_{10}, y_4+y_5</td><td></td><td>y_1+y_2, y_3+y_4</td><td></td><td>y_3+y_4, $y_{10}+y_1$</td><td></td></tr>
<tr><td colspan="11">Step 3 => Retransmit y_3+y_4</td></tr>
<tr><td>Not received</td><td></td><td></td><td>y_4+y_5, $y_{10}+y_1$</td><td></td><td>y_9+y_{10}, y_4+y_5</td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="11">Step 4 => Retransmit y_{4+} y_5 => everything is recovered</td></tr>
</table>

Table 3.1 now shows at step 2 which signals are still missing to recover all other nodes' messages, and a second packet combination is broadcast after consulting the histogram of Figure 3.4 (a).

As packet combinations are progressively re-transmitted, nodes begin to recover all their unknowns – this is shown as empty cells in Table 3.1, taking into consideration that the ten nodes can recover the nine unknown packets even when one packet is not received.

Only four ARQ requests are required via cooperation for all nodes to recover the nine unknown packets instead of every single packet that is not received for all nodes, i.e., ten packets in this example. This results in a significant 60% savings in ARQ transmissions. In fact, in the case where any two combined packets are not received by any source node, a maximum of $N/2$ combined packets is requested to be repeated instead of N uncombined packets when NC is not applied.

The minimum number of ARQ packet requests is one, when one common packet is not received by all N nodes, as in Step 4 in Table 3.1. When three packets are still not received by Nodes 3 and 5, only the y_4+y_5 combined packet is needed, i.e., Node 3 uses y_4+y_5 to retrieve $y_{10}+y_1$ and Node 5 uses the same combined packet to retrieve y_9+y_{10}.

In Table 3.2 another example is shown where different nodes do not receive different number of combined packets and the same process of broadcasting the packets missed by most nodes is applied until all nodes have received N-1 unique packets. We observe that seven packets are requested to be re-transmitted according to Figure 3.4 (b).

Table 3.2 : Seven ARQ requests to the relay when more than two packets not received.

Node	1	2	3	4	5	6	7	8	9	10
Not received	y_3+y_4, y_6+y_7, $y_{10}+y_1$	y_2+y_3, y_5+y_6, y_8+y_9, y_9+y_{10}	y_4+y_5, $y_{10}+y_1$	y_1+y_2, y_3+y_4, y_5+y_6, y_6+y_7, y_7+y_8	y_1+y_2, y_4+y_5, y_7+y_8	y_5+y_6, y_7+y_8	y_1+y_2, y_2+y_3	y_1+y_2, y_4+y_5, y_5+y_6, y_6+y_7, y_8+y_9, $y_{10}+y_1$	y_3+y_4, y_5+y_6, y_8+y_9, $y_{10}+y_1$	y_2+y_3, y_6+y_7, y_9+y_{10}
Step 1 => Retransmit y_5+y_6										
Not received	y_3+y_4, y_6+y_7, $y_{10}+y_1$	y_2+y_3, y_8+y_9, y_9+y_{10}	y_4+y_5, $y_{10}+y_1$	y_1+y_2, y_3+y_4, y_6+y_7, y_7+y_8	y_1+y_2, y_4+y_5, y_7+y_8		y_1+y_2, y_2+y_3	y_1+y_2, y_4+y_5, y_8+y_9, $y_{10}+y_1$	y_3+y_4, y_8+y_9, $y_{10}+y_1$	y_2+y_3, y_6+y_7, y_9+y_{10}
Step 2 => Retransmit $y_{10}+y_1$										
Not received	y_3+y_4, y_6+y_7	y_2+y_3, y_8+y_9, y_9+y_{10}		y_1+y_2, y_3+y_4, y_6+y_7, y_7+y_8	y_1+y_2, y_4+y_5, y_7+y_8		y_1+y_2, y_2+y_3	y_1+y_2, y_4+y_5, y_8+y_9	y_3+y_4, y_8+y_9	y_2+y_3, y_6+y_7, y_9+y_{10}
Step 3 => Retransmit y_1+y_2										
Not received	y_3+y_4, y_6+y_7	y_2+y_3, y_8+y_9, y_9+y_{10}		y_3+y_4, y_6+y_7, y_7+y_8	y_4+y_5, y_7+y_8			y_4+y_5, y_8+y_9	y_3+y_4, y_8+y_9	y_2+y_3, y_6+y_7, y_9+y_{10}
Step 4 => Retransmit y_6+y_7										
Not received		y_2+y_3, y_8+y_9, y_9+y_{10}		y_3+y_4, y_7+y_8	y_4+y_5, y_7+y_8			y_4+y_5, y_8+y_9	y_3+y_4, y_8+y_9	y_2+y_3, y_9+y_{10}
Step 5 => Retransmit y_8+y_9										
Not received		y_2+y_3, y_9+y_{10}		y_3+y_4, y_7+y_8	y_4+y_5, y_7+y_8					y_2+y_3, y_9+y_{10}
Step 6 => Retransmit y_2+y_3										
Not received				y_3+y_4, y_7+y_8	y_4+y_5, y_7+y_8					
Step 7 => Retransmit y_7+y_8=> everything is recovered										

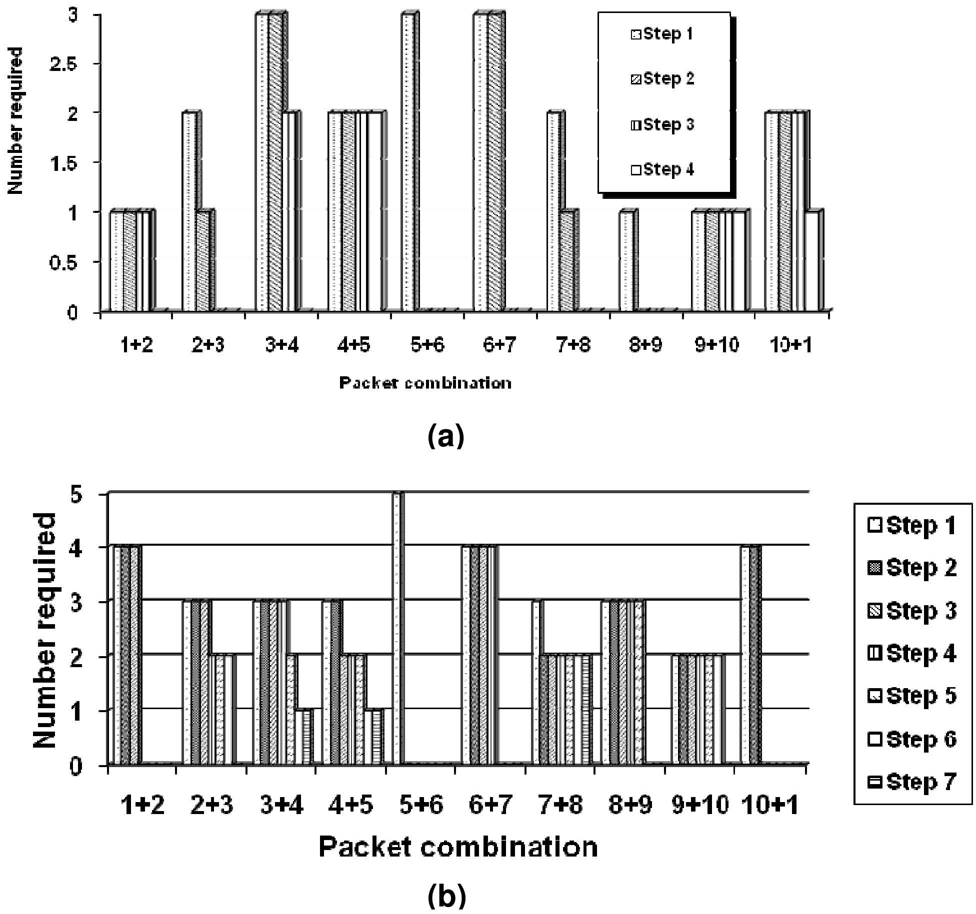


Figure 3.4: Number of packets required for each packet combination missing (a) two (b) more than two packets

In the second example with a worse scenario, only seven combined packets are retransmitted instead of N=10 packets, still resulting in a significant 30% savings in ARQ transmissions. Therefore, through a simple process (at the relay) of counting ARQ requests from all nodes in the network, it is possible to achieve significant savings in retransmissions via cooperative NC.

3.2.5 High-SNR Behaviour

Figure 3.5 shows the AF_b, DF_b/DF_p and AF_p capacities from (3.4), (3.5), and(3.7), respectively vs. SNR in the uplink for two different values of $A=A_{AFb}=A_{AFp}=A_{DFb}=A_{DFp}$. All four schemes use the same total transmitter power.

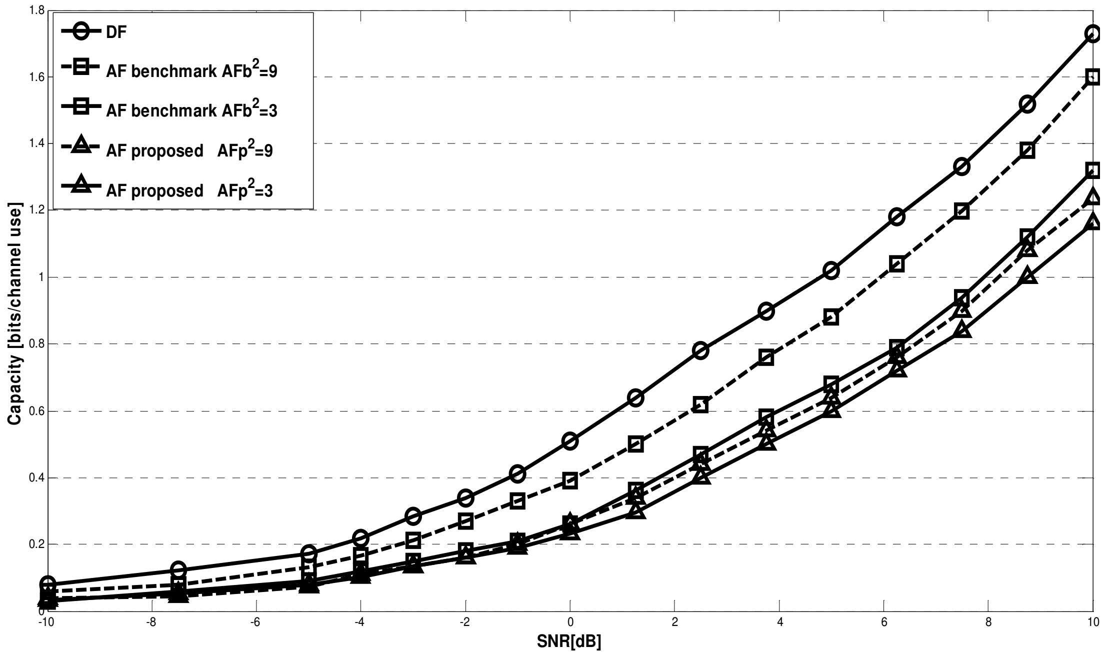


Figure 3.5: The capacities of the four systems as functions of the SNR in the uplink channel.

Since the capacities of both benchmark and proposed DF schemes are identical, they are shown as a single DF curve. It can be seen that DF provides performance gain over AF, which decreases with A.

By comparing dashed and solid lines in Figure 3.5, it can be seen that the gain of DF over AF is less when A is higher, as expected from (3.5) and (3.7). On the other hand, the increase in A enlarges the performance gap between the proposed AF schemes and the benchmark AF. Another conclusion from Figure 3.5 is that, in theory, NC does not provide any capacity improvement when AF is used. This is due to the accumulation of the noises over three separate paths. On the other hand, without NC, each coded message travels via two transmission paths, and thus encounters two independent noisy channels only. However, NC provides savings in the number of transmissions needed.

Two parameters are usually used to measure system performance in the high SNR regime: rate multiplexing gain (or degree of freedom) denoted *r,* and additive gain denoted *a*. Rate multiplexing gain shows how fast the capacity increases with SNR. It is defined by $r = \lim_{SNR\to\infty} \frac{C(SNR)}{\frac{1}{2}\log SNR}$, where C is the capacity.

All four schemes behave similarly in the high SNR regime achieving rate multiplexing gain of one (which is expected since all the systems use a single transmitter antenna). This can be observed from Figure 3.5, as all four curves become parallel.

Additive gain is defined as $a = \lim_{SNR \to \infty} C(SNR) - \frac{r}{2\log SNR}$. It is a shift of the $C(SNR)$ function from the origin at high SNRs. The DF schemes achieve a=0, whereas the benchmark, and the proposed AF cooperative protocol schemes achieve $a = \log \frac{A_{AFb}^2}{A_{AFb}^2 + 1}$, and $a = \log \frac{A_{AFp}^2}{2A_{AFp}^2 + 1}$, respectively. Hence the DF schemes achieve higher additive gain.

3.2.6 PUMTC Behaviour

In this part we show the PUMTC behaviour when soft decision and hard decision decoding processes are used according to

Figure 3.6 and Figure 3.7 that show how PUMTC can decode the received stream of data, decoding at D1 as an example.

In Figure 3.6, we notice that using hard decision results to losing information through the hard decision decoding process. Thence the second design is proposed to avoid this leak of information as shown in Figure 3.7.

It is clear that using soft decision simplifies the system as there is no need to re-encode and modulate the estimated data.

Finally, we could use both designs as a way of error detection, i.e. obtain the same results by the two different designs and then compare the result. In the case that both designs give the same output, we confirm it as final result, but when the result is different, we can consider this as error detection evidence, hence, we re-decode the received data. As our research is aimed at NC, we leave this research point at this level and just show the BER results for both designs.

The results for these two designs are shown in the results Section 3.2.7 next.

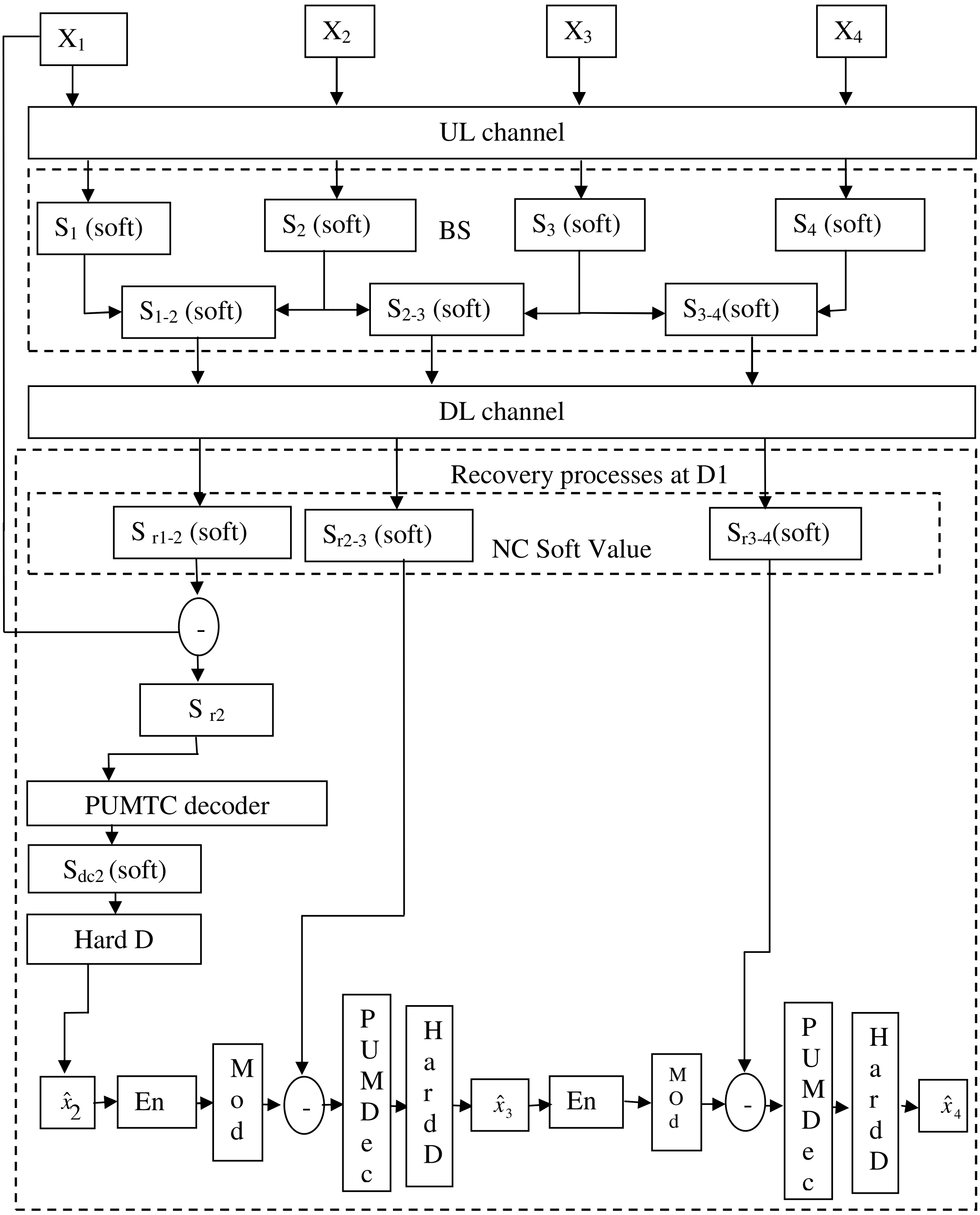


Figure 3.6: Hard decision decoding for Network Coding proposed system for four sources as an example.

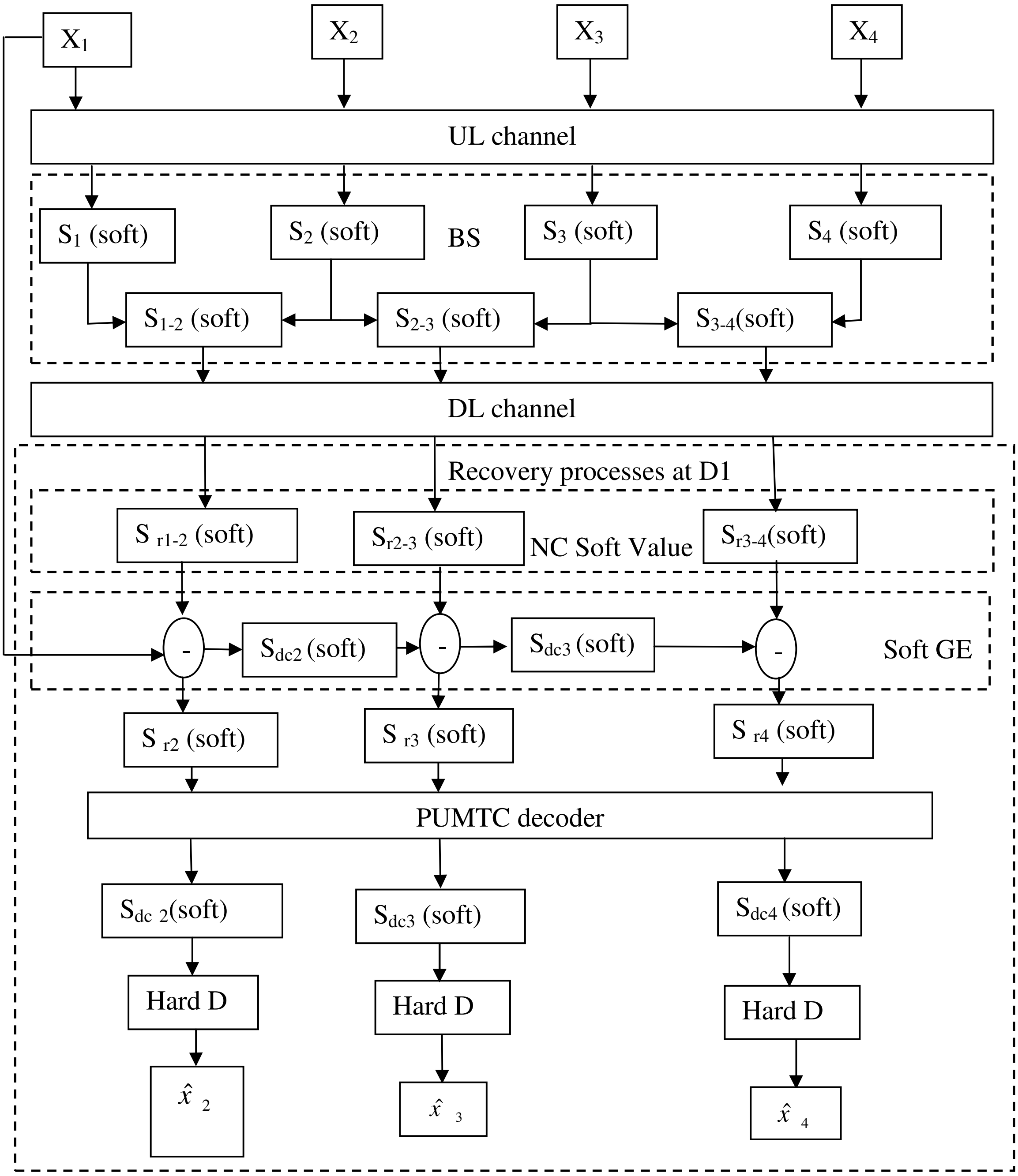


Figure 3.7: Soft decision decoding for Network Coding proposed system for four sources as an example.

3.2.7 Simulation Results

We use Partial Unit Memory Turbo codes (PUMTC) introduced in [87] and showing capacity approaching performance via EXIT charts in [88]. In our systems, transmission is simulated over AWGN, using BPSK modulation for rate 1/3 PUMTCs based on (8,4,3,8) and (4,2,1,4) PUM component codes, and a pseudo-random interleaver of size 1000 bits. The (8,4,3,8) PUMTC is more robust, but more complex than the (4,2,1,4) PUMTC. We set $A_{AFp} = A_{AFb} = A_{DFp} = A_{DFb} = 4$, and four decoding iterations for the simulation run as it gives the best average results. The BER performance curves are obtained by simulating transmission of at least 10^8 bits with at least 100 bit errors for statistical significance.

We start by showing simulation results for AF and DF benchmark and Proposed Systems for just two users, and then extend it to N users.

The results over the physical layer in bit level are shown in term of BER, however, the PER will be introduced in the upper layers research.

We need to confirm that one full packet is received correctly when this packet has error free bits.

Figure 3.8 and Figure 3.9 show the relative BER performance for Two-Source NC in both AF and DF systems for (4,2,1,4)-based and (8,4,3,8)-based PUMTCs, respectively. In Figure 3.8 when the amplification factor (Ap) is increased from 2 to 10, 0.2dB and 0.3dB gains are observed at BER of 10^{-5} for AF_p and DF_p (4,2,1,4)-based PUMTC, respectively. In Figure 3.9, increasing Ap from 2 to 10 results to 0.5 dB and 0.4dB improvement in the performance at BER of 10^{-5} for AF_b and DF_b (8,4,3,8)-based PUMTC, respectively. Moreover, Figure 3.8 and Figure 3.9 show that the DF_b outperforms AF_b by 2dB for (8,4,3,8) compared to 3dB for DF_p over AF_p (4,2,1,4) at BER of 10^{-5}, which is justified by the decoding-and-re-encoding processes performed at the BS, that cancels the uplink noise effect. And finally, it is important to notice that increasing the amplification factor makes no significant improvement, thus transmission power at the BS is kept to an acceptable low value of amplification for the rest of the simulations at four and iteration factor equal to four as well.

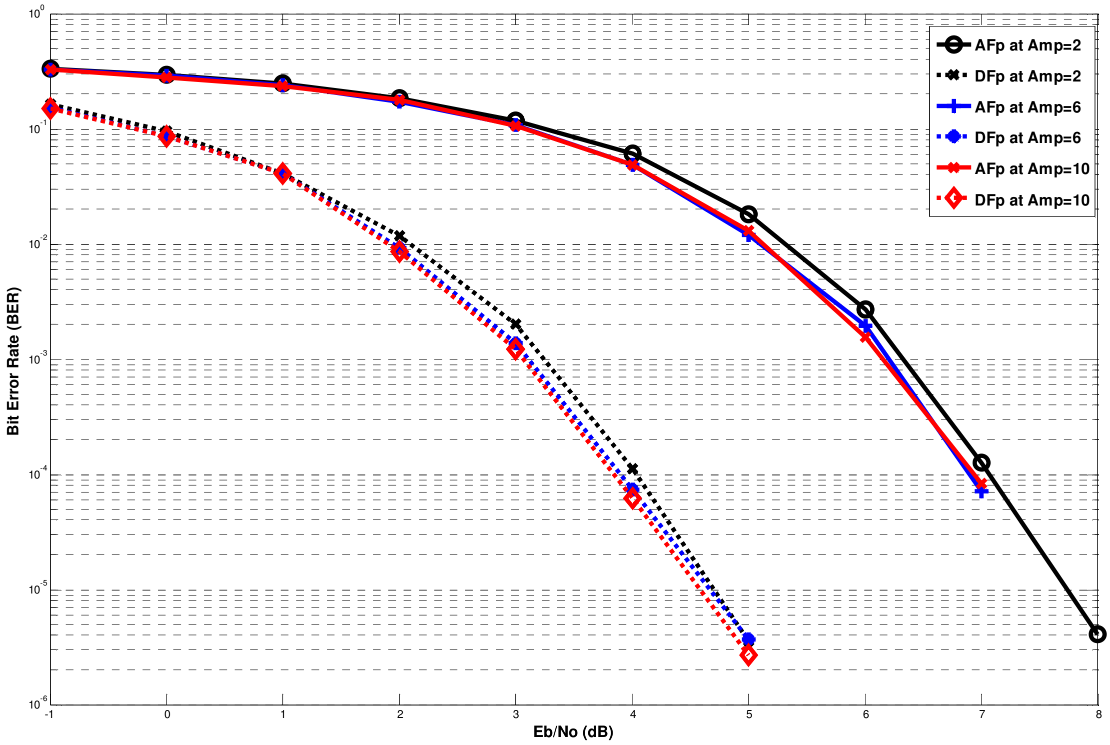


Figure 3.8: AF_p and DF_p systems based on (4,2,1,4)-PUM turbo codes with 4 decoding iterations. The figure demonstrates the effect of increasing the amplification factor.

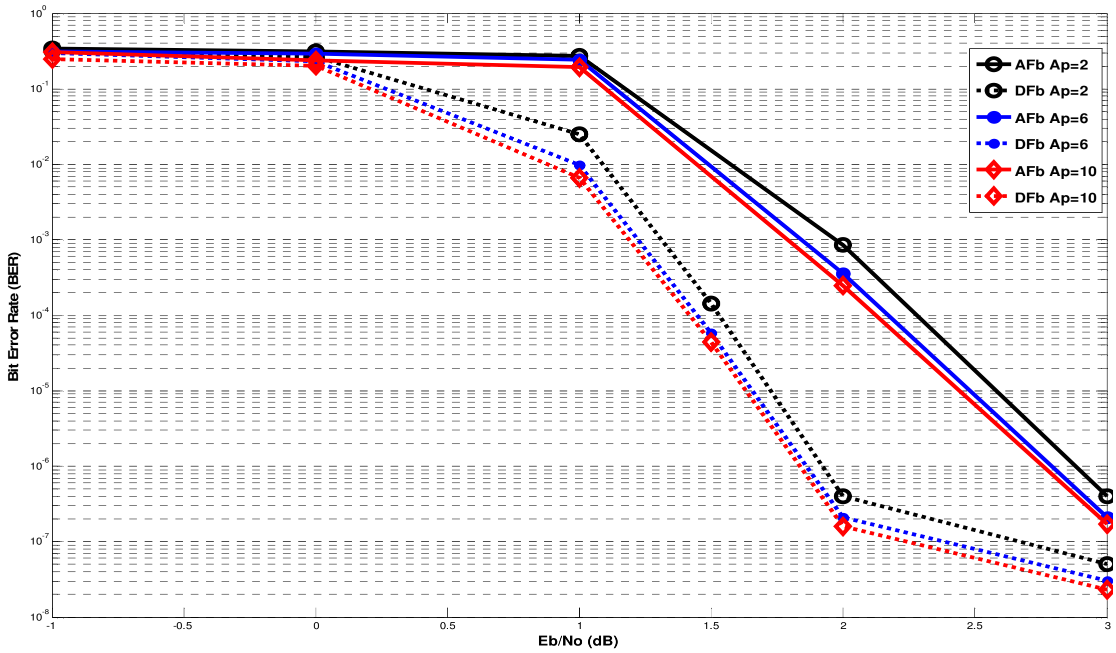


Figure 3.9: AF_b and DF_b systems based on (8,4,3,8)-PUM turbo codes with 4 decoding iterations. The figure demonstrates the effect of increasing the amplification factor.

Figure 3.10 shows the effect of increasing the decoding iteration from 2 iterations to 8 iterations for AF_b and DF_b. PUMTC based on (8,4,3,8) component is chosen as an example. However, systems and PUMTC components behave in similar trend when increasing the iteration, so, it is enough to show one of them as an example.

Moreover, it is clear to notice that here is a distinct improvement in performance as the iteration number is increased. So, the main observation is, at 10^{-5} when the iteration increased from 2 to 8, we gained around 0.9 dB for AF_b, and about 0.6dB for DF_b.

This observation shows how PUMTC is more powerful with AF_b, as just increasing the iteration from 2 to 8 improves the BER by 0.9 dB, than with DF_b, and this is because the of the decoder used at the BS which is as expected.

In addition, it is clear that DF_b outperforms AF_b by 2 dB as a result of the added decoder in the BS.

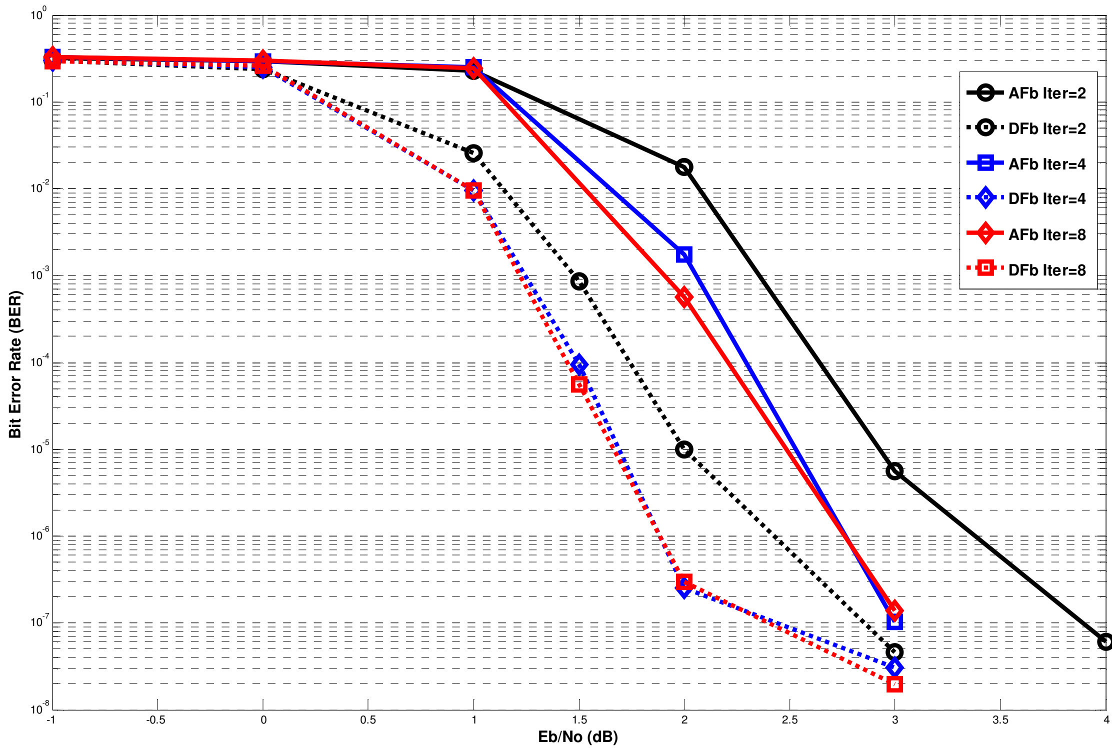


Figure 3.10: AF_b and DF_b systems based on (8,4,3,8)-PUM turbo codes with 4 decoding amplitude. Figure 3.10 demonstrates the effect of increasing the iteration factor.

The results in Figure 3.11 show how PUMTC performs with the proposed AF_p and DF_p for two users compared to their benchmark.

As expected the (8,4,3,8) PUMTC outperforms the (4,2,1,4) PUMTC, and the DF systems outperform the AF systems which are less delay-prone but are noisier. The performance improvement of DF over AF is significantly larger for the (8,4,3,8) PUMTC over the (4,2,1,4) PUMTC, demonstrating the effect in choosing a good channel code. In addition, there is a significant BER performance loss for AF_p compared to AF_b, which is the trade-off in terms of bandwidth savings. On the other hand, the DF_p performance is only marginally worse than the DF_b system, which makes it a better option when performance is more critical and latency is less so.

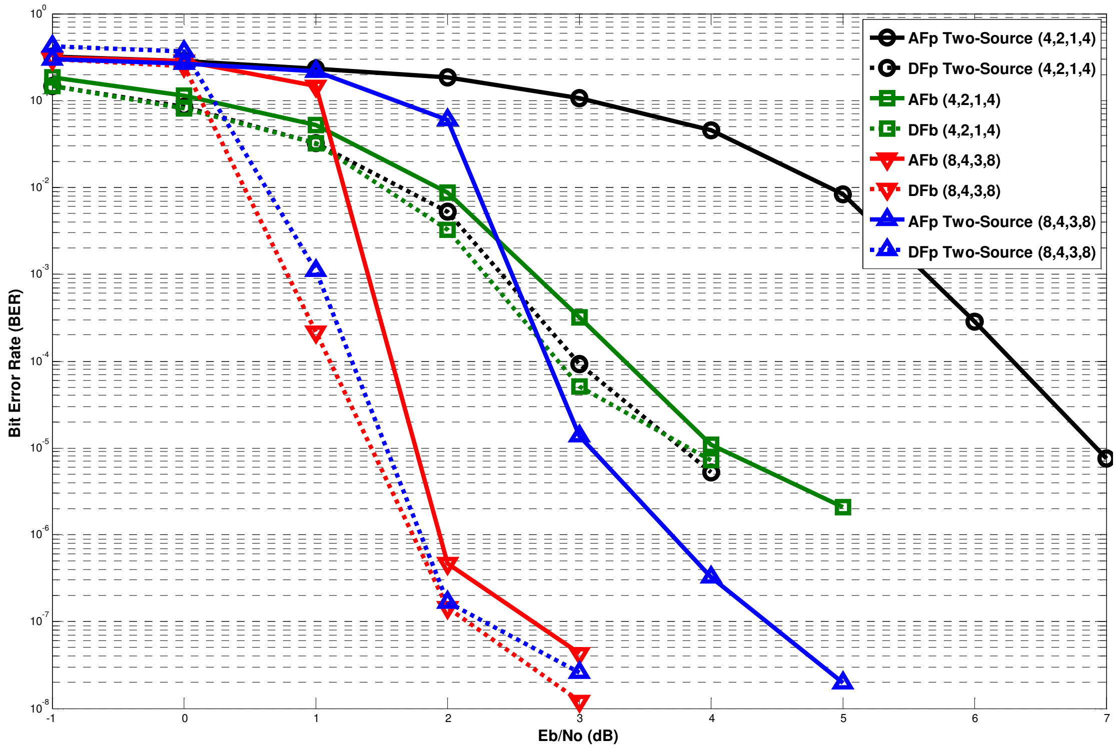


Figure 3.11 : BER for the AF and DF systems based on (8,4,3,8) and (4,2,1,4) PUMTC for N=2.

Figure 3.12 shows the effect of increasing the number of nodes N from 10 to 50 in the network for both proposed AF and DF systems. As expected, with no additional transmitted packets, an acceptable performance degradation of 0.2 dB and 0.3 dB in $AF_p$and $DF_{p,}$ respectively is observed with increasing number of nodes while maintaining only N-1 DL transmissions.

Figure 3.13 demonstrates the influence of increasing the number direct transmitted packets over AF and DF according to (3.13) and (3.14) for AF and DF respectively, showing that sending 99 additional packets (almost double the number of source nodes) to aid in the message recovery process in AF results in a 1.2 dB gain and 1.4 dB in the DF system.

Figure 3.14 shows the decoding performance for the propose AF when N=5. Both soft value and hard values have been performed, and results show that soft value decoding gives better performance, mainly at high BER values, this gap reaches 0.4 dB at BER of 10^{4}, and the gap goes as small as just 0.15 dB at BER of 10^{5}.

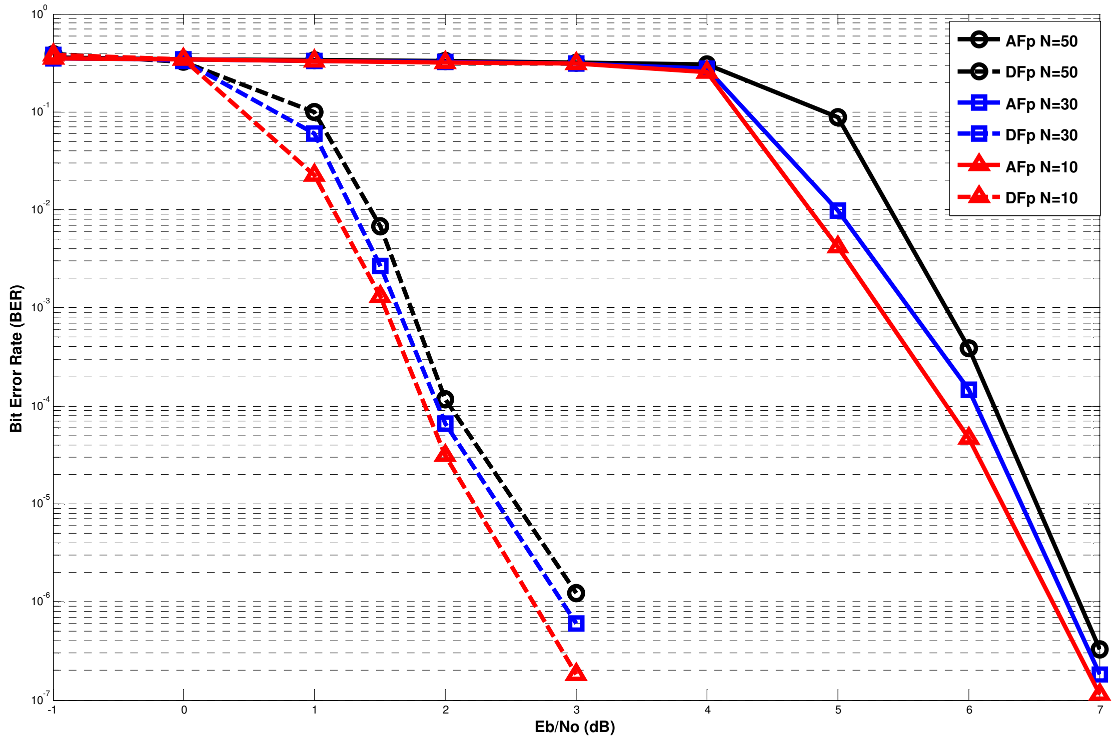


Figure 3.12: AF_p and DF_p systems based on (8,4,3,8) PUMTC for N=10, 30 and 50.

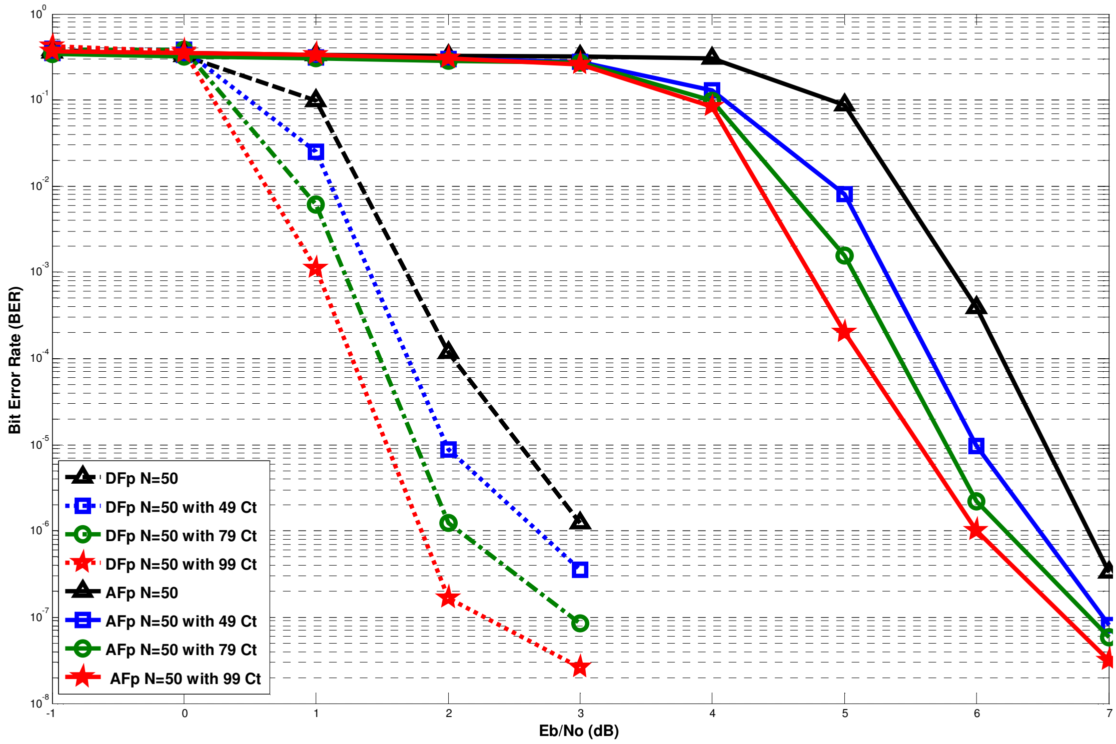


Figure 3.13: AF_p and DF_p systems based on (8,4,3,8) PUMTC, demonstrating the effect of adding up to 99 additional packets.

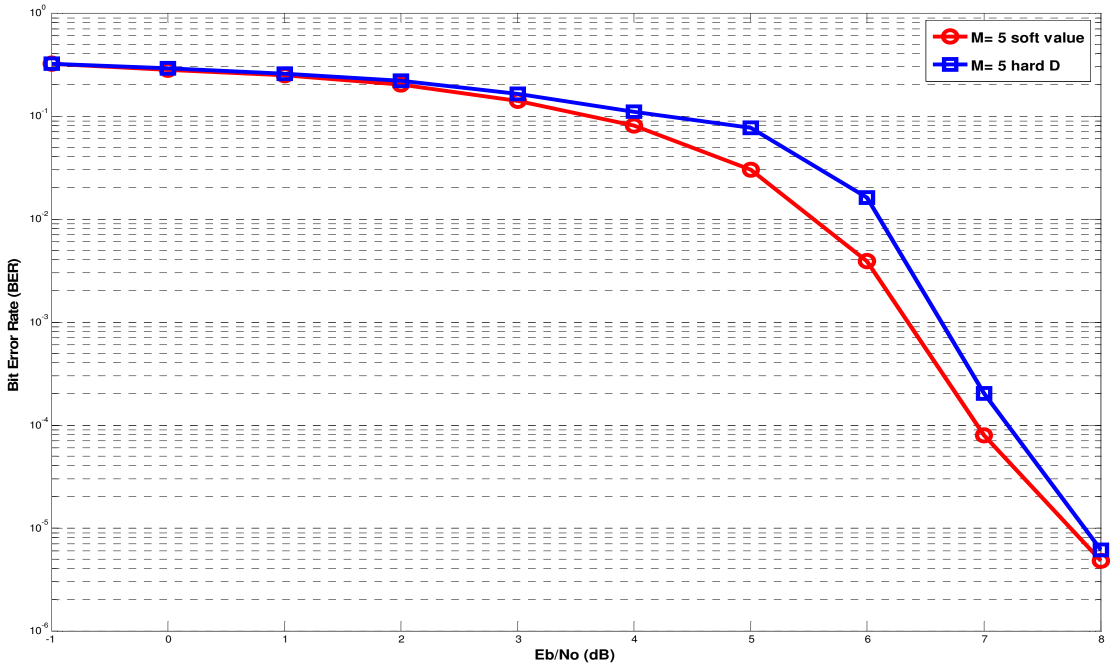


Figure 3.14: BER for Soft and hard decision decoding for AF_p (4,2,1,4) at M=5.

3.3 Physical Layer Network Coding Cooperation Two Stages Transmission

In this section we work with the transmission packet at data stream level using PUMTC as FEC, so, we can claim that the packet is received when all bits stream in this packet are decoded correctly with error free, otherwise, the packet will be identified as not received. However, BER results obtained is for the bit level in term of how many bits are wrongly decoded.

3.3.1 System Set-Up

We consider a WSN that consists of N sources denoted as S1, S2,…, SN that transmit to the same destination D, and to the other sources, via the UL AWGN channel using AF or DF protocols. Assuming that all sources operate on orthogonal channels with the same channel code employed by all sources, so, the N sources and the destination receive sources' dada separately, where all users are in range with each other and to a destination without a relay or a BS in between, and they all aim to receive all data from all neighbours as shown in Figure 3.15.

Figure 3.15 shows the system under investigation for three users and a destination as an example for the benchmark system without applying NC:

In the benchmark scenario shown in Figure 3.15, communication usually occurs in one or two communication stages. First stage, all users broadcast their data to the N-1 neighbours, and then decode the received data using PUMTC as our proposed FEC.

In the case of the failure to decode one or more packet's data at the receiving side, which means in the physical layer, if the user does not manage to decode the received data though using PUMTC as FEC, this user asks for a second transmission stage which is an ARQ for all users to re-broadcast their data and their neighbours' separately without using NC, so, the second stage follows.

In fact, in the second stage, all users will behave as relays to each other and to the destination resulting to N-1 relays for each user and N relays for the destination resulting to broadcasting N packets in the second stage and $N(N+1)$ in the two transmission stages.

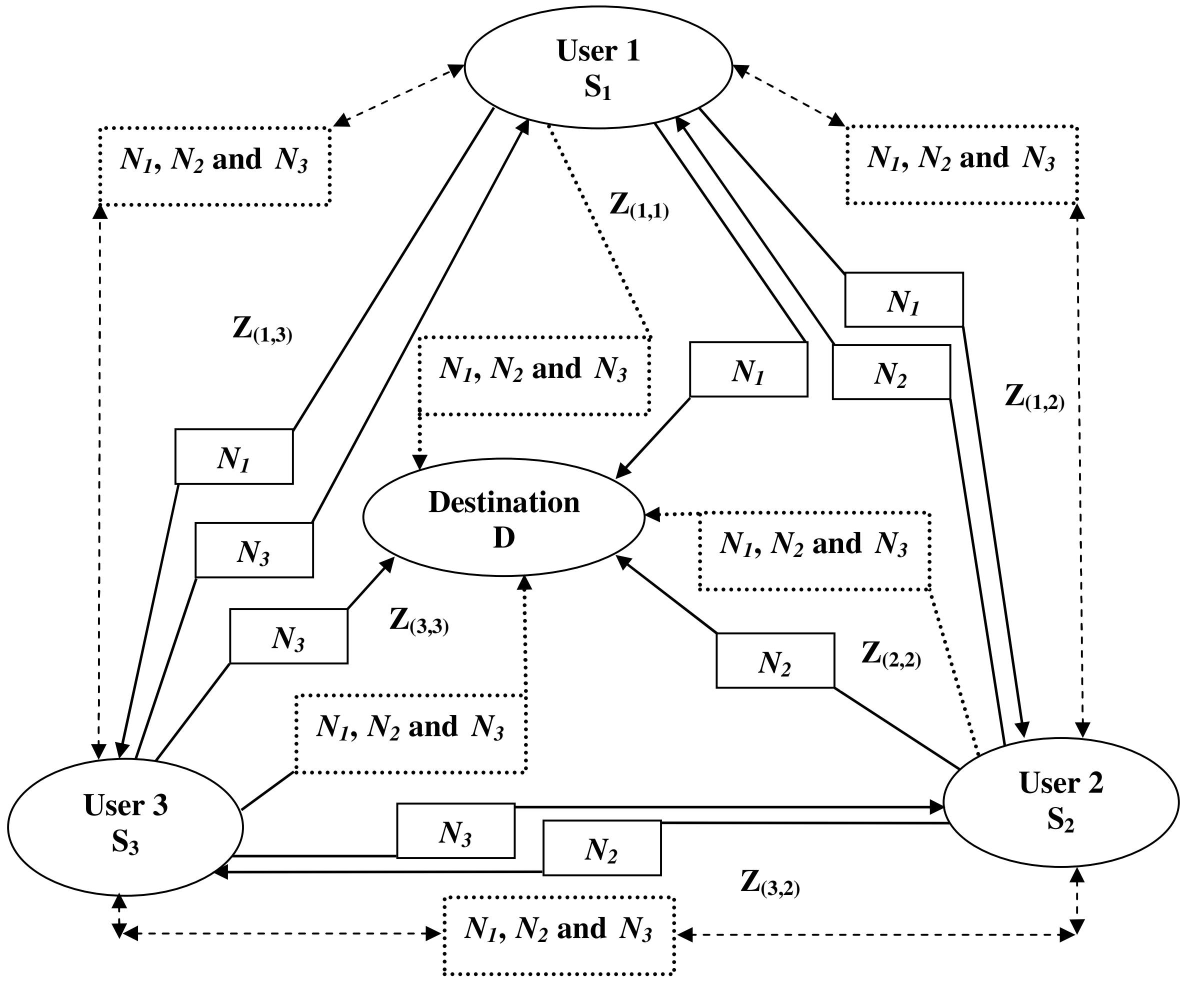


Figure 3.15: Three user's benchmark cooperative system (without NC) for WSN, sold is for first stage and dots for cooperative benchmark second stage.

Moreover, the destination will be receiving a max of $N(N+1)$ at the end of the second stage, and each user will be receiving a max of N^2-1 in term of N-1 in the first stage (solid lines in Figure 3.15) and $N(N-1)$ for the second stage (dots lines in Figure 3.15).

It is clear that the benchmark cooperative second stage suffers from a big number of redundancy transmitted packets resulting to loss in the bandwidth and transmission power.

Our proposed cooperative protocol used in Section 3.3 is expanded in this section to be more practical for this WSN scenario where all users aim to receive all data from all neighbours, and the destination aims to receive all data from N users as well, as shown in Figure 3.15 for three users and a destination.

Deterministic NC mentioned in Section 2.4.2 is proposed in this scenario to improve the communication and obtaining a better throughput.

The main challenge when applying the deterministic combination is to assure fulfilling the essential requirement of having transfer matrices with full rank at each user and the destination [72].

Following we explain the proposed deterministic NC cooperation protocol.

3.3.2 Proposed Network Coding Cooperation Protocol

In our proposed cooperative system where NC is applied, each user generates its packet of binary sequences message, encodes it using PUMTC, amplify and then broadcasts the resulting amplified packet A_pX_i, where i=1,2,…,N and A_p is the amplification factor, directly to the destination and to the N-1 partners at the first stage using N transmission time slots. Equation (3.15) gives the received N packets at the end of the first stage at the destination and the N-1 partners:

$$\begin{bmatrix} \hat{Y}_{(1,1)} & \hat{Y}_{(1,2)} & \hat{Y}_{(1,3)} & .... & \hat{Y}_{(1,N)} \\ \hat{Y}_{(2,1)} & \hat{Y}_{(2,2)} & \hat{Y}_{(2,3)} & .... & \hat{Y}_{(2,N)} \\ . & . & . & .... & . \\ . & . & . & .... & . \\ \hat{Y}_{(N,1)} & \hat{Y}_{(N,2)} & \hat{Y}_{(N,3)} & .... & \hat{Y}_{(N,N)} \end{bmatrix} = \begin{bmatrix} X_1 \\ X_2 \\ . \\ . \\ X_N \end{bmatrix} + \begin{bmatrix} Z_{(1,1)} & Z_{(1,2)} & Z_{(1,3)} & .... & Z_{(1,N)} \\ Z_{(2,1)} & Z_{(2,2)} & Z_{(2,3)} & .... & Z_{(2,N)} \\ . & . & . & .... & . \\ . & . & . & .... & . \\ Z_{(N,1)} & Z_{(N,2)} & Z_{(N,3)} & .... & Z_{(N,N)} \end{bmatrix}, \tag{3.15}$$

where $Z_{(i,j)}$ and $Z_{(i,i)}$ denote to the AWGN in the channel between the ith and jth users and the ith user and destination respectively, and $\hat{Y}_{(i,j)}$ and $\hat{Y}_{(i,i)}$ are the received packets sent from the ith user at the jth user and destination respectively.

Equation (3.15) shows that the capacity for the first stage is as in (3.4).

In the second stage, each user relays the N-1 partner's packets and performs the NC over the N-1 partner's received packets $\hat{Y}_{(i,j)}$ by combining the them, i.e. each user combines N-1 received packets (user's own packet is not included), to broadcast N combined packets Xc_i in N transmission time slots, so, each user and the destination receive the N combined packets over a DL wireless channel as $\hat{Y}c_i$. In such way, a total of 2N transmission time slots is needed for both stages when using Time-Division Multiple Access (TDMA) compared to $N(N$-1) TDMA when NC is not applied.

So, we proposed our cooperative mode to enable the N sources to change data between them all in full-duplex communication set up as well as enabling the destination to retrieve the data received by the N sources using just 2N transmission time slots in TDMA which enhances the robustness significantly.

Following, we explain both the proposed cooperative and selfish modes for AF and DF.

AF Proposed AF_p network cooperative: The proposed network cooperative based on AF system is more suitable for real-time communications since decoding at the relay node is avoided. So, in AF each user amplifies its own packet separately and transmits it directly to the N-1 partners and to the destination in the first stage. In the second stage, the N sources apply NC over the N-1 received packets and then amplify the combined packets before broadcasting.

Fully cooperative combination AF_p: the system behaves fully cooperatively when the N sources manage to receive the N-1 partners' packets successfully, in such case, the N sources apply the NC over the N-1 partners' received packet by adding them together in one combined packet, so, each source combines N-1 packets, i.e., the all received packets without its own packet.

The reason we exclude the user's own packet from the combination is to obtain a transfer matrix with rank N at the destination as all N users which is an essential requirement [72].

Finally, the N sources amplify and broadcast the amplified N combined packets shown in (3.16):

$$Xc_{AF(i)} = A_{AF} \sum_{j=1, i \neq j}^{N} Y_j, \tag{3.16}$$

which are received by the destination and the N-1 sources as shown in (3.17)

$$\hat{Y}c_{AF(i)} = A_{AF} \sum_{i=1, j \neq i}^{N} Y_i + z_{DL(i)} = A_{AF} \sum_{i=1, j \neq i}^{N} X_i + A_{AF} \sum_{i=1, j \neq i}^{N} Z_{UL(i)} + Z_{DL(i)}, \tag{3.17}$$

where i=1,2,…,N, and A_{AF} is the amplification factor.

Equation (3.18) shows that the capacity for this combined packet is:

$$C_{AFc} = \frac{1}{2}\log\left(1 + \frac{A^2 AF_b P_i}{(N-1)A^2 AF_b + 1}\right). \tag{3.18}$$

Equation (3.18) shows that in our proposed AF_p, the capacity decreases severely, i.e., the more packets we combine, the less capacity the system has as a results of the accumulated channel noises.

According to (3.16), the destination receives 2N packets in 2N time slots in both stage one and two compared to N(N-1) in TDMA when NC is not applied.

Selfish operation: when one source fails in receiving one of its partners' packets, it behaves selfishly by retransmitting its own packet again together with a short acknowledgement message to the N-1 partners and destination, this acknowledgement message plays important role in the network decoder as well as in the way to retrieve any missing packet at the destination and N-1 sources. In such case, the system does not work fully cooperatively, however, still the rest of the partners behave cooperatively, so, they combine the N-1 received packets and then broadcast the combined packet as in the fully cooperative mode, and accordingly, we notice that even in the selfish operation mode, the system does not request retransmitting the N packets as in the case of selfish operation when NC is not applied, which keeps the total number of broadcasted packets and transmission time slots equal to 2N for the two stages as in full cooperative mode.

The full selfish operation happens only when all sources have problem with receiving or decoding one or more packets. In such case, each user sends its own packet just once separately together with the acknowledge message, which means that the system is most of the time working as cooperative or partly cooperative mode, and in the case of selfish, just N time slots are needed to re-transmit the N packets compared to $N(N\text{-}1)$ in TDMA when NC is not applied.

DF Proposed DFP Network Cooperative: The proposed network cooperative based on DF system is more suitable for time-delay communications since time is needed for the decoding at the relay nodes and destination. So, in DF each user decodes and re-encodes its received partners' packets, amplifies them with its own packet separately and transmits the N resulting packets directly to the N-1 partners and the destination in the first stage. In the second stage, the N sources apply NC over the N-1 re-encoded received packets and then amplify the combined packets before broadcasting.

Fully cooperative DFP: the system behaves fully cooperatively when the N sources successfully decode the received N-1 partners' packets. In such case, the N sources apply the NC over the N-1 partners' received packet by adding them together in one combined packet using modulo-2 $\oplus$, so, each source combines N-1 packets, i.e., the all received packets without its own packet.

Finally, the N sources amplify and broadcast the amplified N combined packets as (3.19):

$$Xc_{DF(i)} = A_{DF} \sum_{j=1, i\neq j}^{N} \hat{X}_j = \left[\hat{X}_1 \oplus \hat{X}_2 \oplus \ldots . \oplus \hat{X}_N \oplus X_i\right] \tag{3.19}$$

which are received by the destination and the N-1 sources as shown in (3.20)

$$\hat{Y}c_{DF(i)} = A_{DF} \sum_{j=1, i\neq j}^{M} \hat{X}_j + Z_{DL} \tag{3.20}$$

where i=1,2,…,N and A_{DF} is the amplification factor.

Figure 3.16 shows the two transmission stages for the fully cooperative NC coded protocol.

Equation (3.20) shows that the capacity for this proposed DF cooperative protocol is as in DF_b and DF_o (3.6) or (3.9). This means that, unlike AF_p, increasing the number of combined packets does not decrease the capacity.

When comparing (3.6) or (3.9) with (3.18), we notice that our protocol fits DF system, which is the reason we restricted our results part to deal with DF protocols.

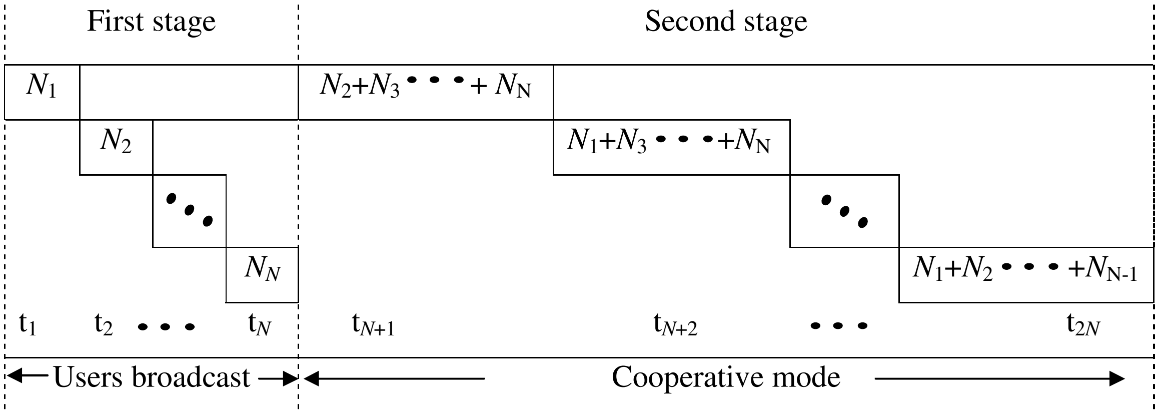


Figure 3.16: Network coding over fully cooperative protocol, stage one and two are shown.

According to (3.20), the destination receives $2N$ packets in $2N$ time slots in both stage one and two compared to $N(N$-1) in TDMA when NC is not applied.

DF$_P$ selfish operation: when one source fails in decoding one of its partners' packets, it behaves selfishly by retransmitting its own packet again together with a short acknowledgement message to the N-1 partners and destination, as in AF$_p$ selfish operation. In such case, the system behaves exactly like AF$_p$ selfish by retransmitting just only the packet for the source which fails to decode one or more partner's packet. However, the rest of the partners behave cooperatively, so, they combine the N-1 re-encoded packets and then broadcast them to the destination as in the fully cooperative mode, and accordingly, as in AF$_p$ selfish operation mode, the system retransmits just the packets of the sources which fail to decode one or more partners' packet. So the total number of broadcasted packets and transmission time slots equal to $2N$ for the two stages as in full cooperative mode and AF$_p$.

The full selfish operation happens only when all sources fail in decoding one or more partners' packets, and in such case, all sources send their own packets just once separately together with the acknowledge message. So, in the case of selfish operation mode, just N time slots is needed to re-transmit the N packets compared to $N(N$-1) in TDMA when NC is not applied.

Thus, a 1-bit header is included for Stage 2 to inform D whether the packet is combined or not. Since combination is deterministic, no additional header information about the combination pattern must be sent in order for D to recover all X_i. Only one packet from each source is sent to D and to the *N* users *N* time slots. And it is always assured that the transmitted matrix has full rank at the destination and the *N* users.

3.3.3 Network Decoding for the Cooperative Network

At the receiving side, both the destination and the *N* users will have a maximum of 2*N* unique packets, *N* unique packets as uncombined packets $\hat{Y}_i$ shown in (3.15), and the other *N* unique packets as combined packets shown in (3.17) $\hat{Y}c_i$ for AF and (3.20) $\hat{X}c_i$ for DF.

As in traditional wireless NC, in order to recover all *N* X_is, a minimum of *N* linearly independent equations are needed at D and *N*-1 at each user (as the user's own packet is known at its side), i. e., the received matrix must have full rank at the destination and the *N* users.

Gaussian elimination is applied to recover all packets soon D and the *N* users receives the *N* linearly independent equations. The deterministic combination adapted in our cooperative stage has several characteristics regarding obtaining the full rank transmitted matrix; we briefly explain them bellow and show the benefits behind them.

The most important feature of our combination is that all users and D can recover the *N* packets even if D or any user does not hear from one user (neither the packet sent in stage one nor the packet sent in stage two are received).

The reason why it is possible to do so is the fact that it is enough to receive *N* unique linearly independent equations to recover the *N* unknown packets and, as we are combining the packets in the second stage in a linear deterministic way, it is guaranteed that this un-received packet will be included in all of the combined packs excepts the uses' combined packet, as shown in (3.17) and (3.20) for AF or DF respectively, so, this packet is being received indirectly through all the other user's combined packets, which make it possible to be retrieved, taking into consideration the fact that it is essential to received full rank matrices at D and the *N* users.

In addition, it is enough for D or the *N* users to receive any *N*+1 out of the 2*N* unique packets to recover all *N* packets, taking into consideration that the users *N* know their own packets, so, they need more *N* packets.

This is because the N deterministic combined packets from the second stage give N-1 linearly independence equations. Any other packet from stage one will be a new linearly independent unique equation to be added to the N-1 linearly independent unique received in the second stage. In addition, the N equations sent in stage one are all linearly independent as shown in equation (3.15), so, any N+1 mixed unique equations from stage one and two gives N linearly independent equations.

Finally, it is important to mention that it is impossible to recover the N packets at D or any users if two users are not heard at D or the N users. This is because if two users are silent, the rest of the packets (2N-4) give N-1 linearly independent equations; hence, it is impossible to obtain the transfer matrix with full rank to recover all N packets. Third stage of transmission or AQR is needed to solve this problem.

At the destination side and the N users, the retrieving process starts with the received direct packets, and in the case of not received (AF) or decoded (DF), the destination resorts to combined packets to retrieve the requested packet(s), with no need to request the re-transmission.

Accordingly, the number of retrieving steps depends on the number of packets received in the first stage, in term of the more packets received in the first stage, the fewer retrieving steps are needed. Moreover, the more retrieving steps are needed the more BER aggregation as shown in the results part in Section 3.2.7.

3.3.4 BER Results for Network Coding Cooperation Protocol

Transmission is simulated over AWGN channel, using binary phase shift keying (BPSK) modulation for rate 1/3 PUMTCs based on (8,4,3,8) PUM component codes, a pseudo-random interleaver of size 1000 bits is used. These PUM code designs and turbo code set-ups with amplification factor equal to 4 and iteration equal to 4. The BER performance curves are obtained by simulating transmission of at least 10^8 bits and at least 100 frame errors are guaranteed to be collected for statistical significance.

We ran the simulation for different number of users N=3, 5, 10 and 15, and in each case we assume that one direct transmission is not received through the first stage, and this packet has been retrieved by the cooperative combined packet received from the second stage, without the need to repeat

Figure 3.17 shows the BER for the retrieved packet which is not received directly.

Figure 3.17 shows that the BER for the direct transmission in stage one gives the best BER outperforming the transmission through a relay when NC is not applied by around 0.1 dB, which reflects how the decoding and re-encoding is powerful when used in both the destination and the relay. When NC is applied in our proposed model, we can see that there is loss of around 0.2 dB with compared with no NC applied, which is justified by combining *N*-1 packets in the *N* relays (partners). Moreover, we notice that increasing the number of the combined packets results to more loss in the BER though it is less that 0.2 dB when increasing the number of combined packets from 2 to 14 packets, which is expected as the decoding and re-encoding removes the UL noise.

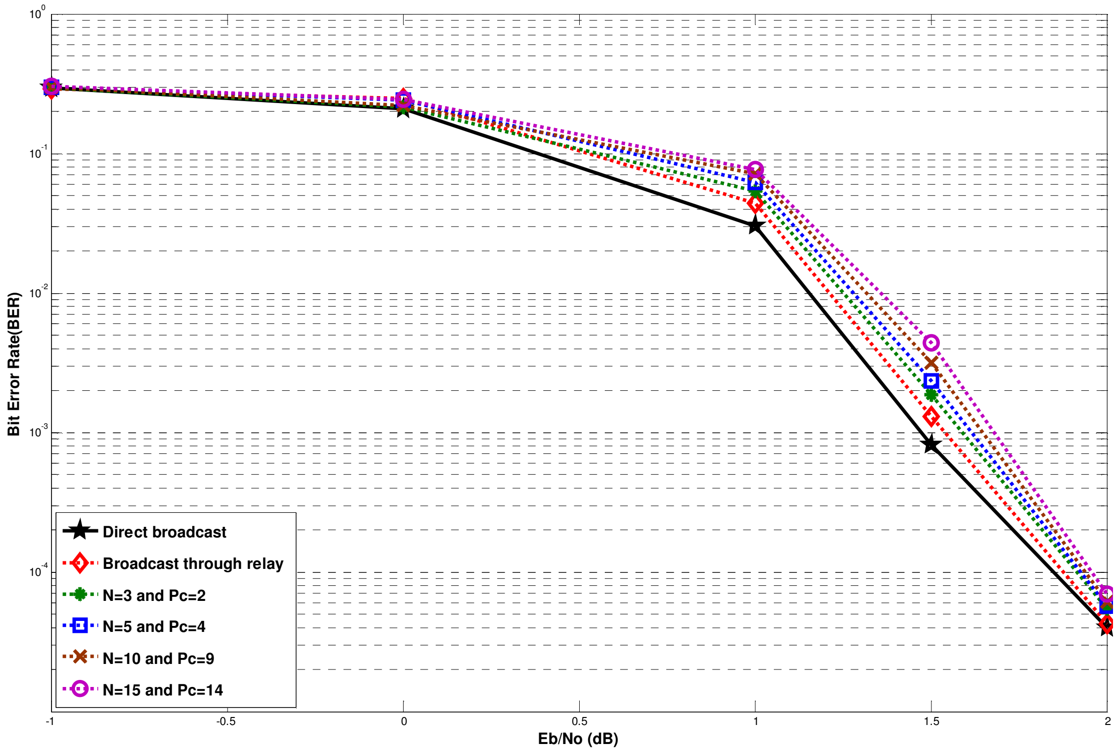


Figure 3.17: DF BER for the retrieved packet which is not received in first stage for *N*=3, 5, 10 and 15 compared to direct transmission and through a relay without applying NC.

Finally, we need to remind that the results in Figure 3.17 obtained by using the direct transmitted decoded packets to find out any requested packet through the N combined packets in case of one directed transmitted is not received or decoded at the destination, which means that we still have the same robustness obtained when NC is not applied but with just $2N$ transmitted packets instead of N^2 when NC is not applied.

3.4 BER and the Number of Combined Packets Trade-off System Mode

In stage one, each of the N users receives N-1 packet and the destination receives N packets as in equation (3.15).

Prior to stage two, each user applies NC in cooperative mode, i.e. combination of all N-1 if all previously received and then relays the packet or just retransmits its own packet (selfish mode).

So, after stage two, the destination and N users receives a max of N noisy packets $\hat{Y}_i$, i.e., (3.17) and (3.20) for AF_P and DF_P respectively.

According to (3.17) and (3.20), the more packets we combine at each source, the noisier the broadcast packets in stage two, mainly in AF protocol where the BS (neighbours) does not perform any decoding. We can improve the BER at the destination and the N users by combining fewer than N-1 packets at each source using deterministic or non-deterministic combinations. For example, combining only 2 packets at each user gives the best BER case scenario, as the retrieval process at the decoder is simplified to one process step which minimizes the error propagation, and the capacity of the channel will be acceptable as shown in (3.6). This comes at a cost: increased transmission bandwidth or transmission of the combined packets in more time slots.

So, in AF there is a trade-off between transmission bandwidth, BER and channel capacity, the main trade-off in such design is clearly the data rate as we need $(N\text{-}1)/P_c$ combined packets in the second stage, where P_c is the number of user's combined packets transmitted per time slot for each user. For example, when P_c= 2, each user combines just two packets, which gives the worst data rate when applying NC with the user cooperative mode but it achieves the best BER and channel capacity. However, it is still as twice as better data rate when NC is not applied for TDMA as (N-1) broadcasted packets are needed.

Moreover, it is the same for the transmission time slots, as (N-1)/P_c time slots are needed for the second stage compared to (N-1) when NC is not applied.

The outcome of the sources cooperative, is that the system's design is being chosen mainly according to the transmission time slots and the BER and the main way to trade-off between these two factors is the number of the combined packets P_c as the all of the number of the transmission packets, transmission time slots and the noise components depend on P_c.

With DF, increasing P_c results to minimum BER effect as shown in Figure 3.17, with improving the data rate significantly at no capacity loss as shown in (3.7), the main trade-off in DF is the time delay, the higher number of combined packets the longer retrieving time needed.

In conclusion, we can generalize the trade-off to be small values of P_c is recommended to be for AF, and high value of P_c is recommended to be for DF with taking into consideration the time delay.

Results show the trade-off mainly at high number of P_c for DF, and the loss in capacity for AF in the part of capacity analysis in (3.18).

3.5 Conclusion

This chapter considers a low-complexity physical layer network encoding and decoding scheme for bandwidth- and power- savings for an information exchange scenario via a relay with Amplify-and-Forward (AF) or Decode-and-Forward (DF) based schemes. The systems combine NC with high-performance partial-unit memory-based turbo codes for forward error correction. The theoretical limits of capacity for the proposed schemes are shown with the schemes' behaviour in high and low SNR regimes. We propose a deterministic combination scheme where messages from two nodes are combined by the relay before broadcasting by AF or DF, yielding a saving in 1/N in transmissions. A modified version of the Gauss-Jordan elimination algorithm is proposed for message recovery exploiting the system set-up. We propose broadcasting additional packet combinations to decrease the effect of error propagation inherent in the recovery process. Simulation results for all proposed schemes demonstrate their relative performance over the benchmark scheme, and are promising due to their performance and simplicity.

Additionally, we propose a cooperative NC scheme to reduce transmissions of ARQ packets when source nodes across the network do not receive packets. Again, we show that when the relay broadcasts most requested combinations in a step-by-step fashion, it is possible to achieve significant savings in transmissions over traditional (benchmark without NC) transmission of uncombined packets.

Moreover, deterministic NC has been provided to be applied over cooperative scenario of two stage transmission and then compared with the benchmark cooperative scenario, showing how NC is useful and it does improve the throughput.

In addition, the trade-off between the data rate and capacity has been shown for both AF and DF.

Finally, the simulation results for all proposed schemes demonstrate their relative performance over the benchmark scheme, and are promising due to their performance and simplicity together with the capacity theoretical limits for all schemes.

4. Cooperative Network Coding for erasure channels, Uplink and Downlink Scenarios

4.1 Introduction

The design and analysis of wireless cooperative transmission protocols have gained significant recent attention to improve diversity over lossy and fading channels. The fundamental idea of cooperation, rooted in the seminal paper of Cover and El Gamal [99], is that users within a network transmit their own information, and serve as relays to help other users' transmissions. The achievable gains for relay channels are shown in [100]. Several energy-efficient cooperative transmission schemes have been investigated in [101] characterizing their outage behaviour. Cooperation via distributed channel codes across orthogonal user channels became popular because it was simple, effective and practical [10] and [103]. A survey of recent information-theoretic results and coding solutions for cooperative wireless communications is given in [83]. Most existing cooperative protocols operate in a timesharing (or frequency-sharing) manner, such that each user sends its own messages and relays its partners' messages in separate time slots (or frequency bands). In order to improve system throughput, it is possible to combine the users' messages. For example, in [104], each user transmits a linear combination of its own waveform and the noisy waveforms received from its partners resulting in significant bandwidth savings at the cost of increased decoding complexity. In [105] and [106], cooperation is achieved through NC [107] and [108] when each user transmits an algebraic superposition of its own and its partners' information, and decoding at the destination is carried out by iterating between codewords from the two users. These schemes provide substantial coding gain over cooperative diversity techniques without NC, motivating the use of cooperation via NC.

This chapter proposes and analyses simple packet-level cooperative transmission protocols that increase efficiency of multiple Wireless Sensor Network (WSN) nodes in delivery of data to a joint central point -destination node - through the use of NC. In contrast to [104], [105] and [106], where physical layer NC is investigated, we consider NC over equal-length packets implemented as a sub-layer on top of the link/MAC layer, as proposed in [109]. Aiming for a low complexity, practical solution suitable for localized (cluster) WSN node cooperation with small header requirements, we resort to NC over a binary field (i.e., simple bit XOR-ing). We allow for packet losses over wireless block fading channels between users and each user and the destination.

We start with a simple design of an erasure channel, assuming that all users to users channel with the same erasure probability, regardless the distances, and the same for all users to the destination, and then, full analysis is performed.

Next, we are applying NC cooperative over the downlink communication to prove that NC cooperative technique is as useful in the uplink.

The NC cooperative scheme has been applied over LTE-A scenario where the Pico relay receives the data from the source and forwards them to the Femto relays, whose then forward them to the users (downlink scenario).

Accordingly, we show our proposed work as a result of applying Cooperative NC over the erasure channel in Section 4.2. Section 4.3 shows the uplink data gathering scenario with the proposed protocols for this scenario and the probability analysis and results. Section 4.4 shows the downlink scenario for the erasure channel, when applying NC over LET-A one Pico and two Femto relays cooperative in Section 4.5. And then Section 4.6 shows the results for both uplink and down link scenarios. And finally, Section 4.7 gives the conclusion for this Chapter.

4.2 Erasure Channel Cooperative System

We design and analyze two complete systems, the first one enables cooperation among multiple users through the use of NC over the uplink communication in the erasure channel. The second system is directed to LTE-A scenario where a Pico relay forwards the packets to two Femto relays; and then from the Femto relays to the users, i. e., applying NC over the downlink communication. Accordingly, the first system scenario allows the NC to be performed over the users which behave as relays, while in the second scenario NC is performed over the BS. The advantages of the two systems are their simplicity while still maintaining performance gains over non-network coding solutions. We allow for packet losses in the erasure channel, assuming that the erasure probability is fixed between users. Moreover, we consider the erasure probability between the users and the destination to be fixed as well; then we analysis the probability to recover all users' messages at a common destination. So, we have two different scenarios, as the first one is an uplink scenario for data gathering, and the second one is a downlink scenario for broadcasting data. In both scenarios, destination receives a set of linear combinations of users' data in two stages in the first scenario or directly from the Pico and relayed by the two Femtos in the second scenario, with an optional third stage in case of failure to decode some messages in the previous two stages in the first scenario or more Femto relays in the second scenario.

4.3 Cooperative System in the Uplink Data Gathering Scenario

In this scenario, a user operates in cooperative mode when it forwards a network coded packet, i.e. a linear combination of its partners' packets.

Aiming for a low complexity, practical solution with very small header requirements, we resort to NC over a binary field. We derive analytical expressions for probability of packet error at the destination for several different low complexity protocols with and without binary NC.

4.3.1 System Model for Erasure Channel

We consider a wireless network that consists of M users $N_1, N_{2,\dots,} N_M$ which transmit messages to a common destination D, and can hear one another over orthogonal erasure channels.

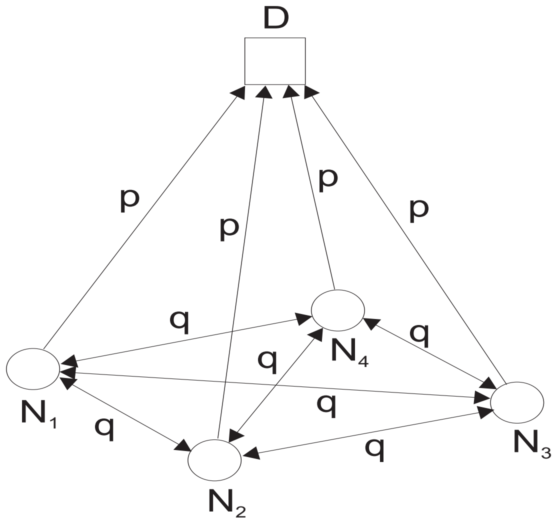


Figure 4.1: System Model with M=4 Users and Destination D.

We assume that the channel between any two users can be modeled as a random packet erasure channel with probability of packet loss q, and the channel between any source and destination is modeled as a packet erasure channel with a probability p of packet loss. This is shown in Figure 4.1.

Communication is done in two or three broadcasting stages of M time slots each, using TDMA. Note that the results hold for other multiple access techniques, including FDMA. User N_i generates at each stage a binary packet which is broadcasted during the i^{th} time slot. All packets are of the same length. For simplicity we assume no packet erasure coding. The destination informs the users with one bit broadcast feedback message when it successfully decodes all M messages. We assume that this feedback message is always perfectly transmitted.

In the first transmission stage, each user N_i broadcasts its own packet X_i in one time slot. Each of the M users receives a packet with probability $1-q$ from the other M-1 users. The destination D receives a packet from each of the M users with probability $1-p$. Stage 1 ends after M time slots.

In the second transmission stage, we propose deterministic and non-deterministic cooperative combination strategies with their relative merits and disadvantages as discussed in the subsections below. Our proposed combination strategies can be seen as simple NC operation over binary field. Indeed, as in NC, a node computes a linear combination of incoming packets. However, in traditional NC [64], each packet is multiplied by a random coefficient, and all multiplicative coefficients are sent in the header. In our setup, all multiplicative coefficients are 1, and hence there is no need for additional header information.

Choice of the best combination strategy depends on the losses encountered. The main advantage of deterministic combinations over their non-deterministic counterparts is that no header information is needed to inform *D* which packets were combined. Only one combined packet (of the same size as the individual packets X_i) from each user is sent to *D*. Thus, Stage 2 requires only *M* transmission time slots. Note that, if the packets were not combined, and each cooperative user still had to send *M*-1 packets belonging to its partners, then transmission would occur in *M(M-1)* time slots. Moreover, when using deterministic NC, there will not be any repetition in transmitted packets as all the generated packets are guaranteed to be unique.

4.3.1.1 Stage 2 M-1 Packet Deterministic Combination

If all *M* users successfully recover their *M-1* partners' packets sent from Stage 1, the system switches to fully cooperative mode, whereby user N_i combines packets received from all other users except its own, as given in (4.1):

$$C_{r(i)} = \sum_{j=1, j\neq i}^{M} X_j, \tag{4.1}$$

However, if any user fails to receive any one of its partners' packets from Stage 1, only that user behaves selfishly by retransmitting only its own packet without resorting to(4.1). Thus, the network will remain in fully selfish operation only when all *M* users have not received one or more packets from their partners. In situations where such losses on such a large scale are likely to occur, in order to fall back to selfish network operation, we propose to combine only two packets either in deterministic or non-deterministic way as described next.

4.3.1.2 Stage 2 2-Packet Deterministic Combination Next Neighbour

This protocol is ideal when any user N_i is very likely to receive from stage 1 its nearest neighbour's packet but unlikely to hear from all other M-2 users as shown in (4.2):

$$C_{r(i)} = \sum_{j=i-1}^{i+1} X_j, \tag{4.2}$$

In this protocol, the user transmits cooperatively in stage 2 by combining its own packet X_i with either X_{i+1} or X_{i-1}. Addressing is cyclic such that if $i=M$, then $i+1=1$.

Of course, if any user N_i does not receive its immediate neighbour's packet, it operates in selfish mode transmitting only its own packet in stage 2.

4.3.1.3 Stage 2 2-Packet Non-Deterministic Combination

This protocol is a variation of the above deterministic protocol described above, but ensures a higher likelihood of more users transmitting cooperatively in stage 2. For this strategy, any user N_i combines its own packet X_i with any one other user's packet X_j it has received. Simulation results show that instead of finding an X_j randomly from the buffer of received packets from stage 1, performance is improved if X_j is chosen when the difference $|i-j|$ is minimum for all possible js. If the difference is 1, the strategy boils down to the above strategy of subsection 4.3.1.2.

Combination of two packets at each user in such a non-deterministic way requires a required header with log_2M bits, as D needs to know j.

4.3.1.4 Stage 2 All-Received Non-Deterministic Combination

Whilst the previous two combination strategies ensure that more users operate in cooperative mode than in the strategy of subsection 4.3.1.1, they limit the possibility of recovery of all M users' messages at D with as little as possible number of received packets k. We thus propose another non-deterministic combination strategy where user N_i combines its own packet with all other packets received from other users, such that:

$$C_{r(i)} = \sum_{j=1}^{M} X_j, \tag{4.3}$$

where X_j is zero if X_j is not received at N_i. If all users' packets are received by N_i, $C_{r(i)}$ is made up of the sum (in binary field) of all M users' packets. This strategy also boils down to the strategy of subsection 4.3.1.6 to subsection 4.3.1.7 if only one other packet is received by N_i.

Due to its non-deterministic nature, this strategy requires an M-bit header listing all j's combined.

After Stage 2, a total of $2M$ unique equations are generated by the users. A maximum of $2M$ unique equations is received at D when and only when all users operate in cooperative mode. The more users operating in selfish mode, the less unique equations are obtained. Due to transmission over a lossy network, only $k \leq 2M$ unique packets (unique equations) will be received by D.

As in traditional wireless NC, in order to recover all M X_i's, a minimum of M linearly independent equations are needed at D. Since operations are done in the binary field and all equations either reveal one unknown or contain binary sums of $M-1$ unknowns, it is enough for D to receive at least $M+1$ unique packets from the M users (either in selfish or cooperative mode) to obtain M linearly independent equations. Under such case, D can recover all sources even if it has not received both packets (sent during both stages) from *one user*. In fact, in many cases, D can even recover all M X_is packets with M unique packets received under many cases as will be shown in Section 4.3.2.

We go to the third stage of transmission if D fails to receive at least M linearly independent equations from M or M+1 unique packet, because it is impossible for D to recover all M source messages. This can happen when more than one user is not heard by D at both stages, or if there are one or more dead users. A user is considered dead if it cannot be heard by D and other users in both stages. The third stage should provide novel information to D in the form of linear combinations not previously received at D. We propose deterministic and non deterministic combination strategies for stage 3 next. This stage also requires only M time slots for transmission. D can recover the M users' messages when M linearly independent equations are received from all three stages. At this point, only if there are dead sources in the network will D not be able to recover all M users' messages.

4.3.1.5 Stage 3 M/2 Odd-Even Deterministic Combination

In this protocol, $M/2$ packets are combined at each user, resulting in a simple, deterministic solution. This is shown in (4.4), where odd numbered users N_i combine only all received packets from their odd-numbered partners excluding their own packet. The algorithm is symmetric for even-numbered users.

$$C'_{r(i)} = \begin{cases} \sum_{j=1, j\neq i,\ j\%2\neq 0}^{M} X_j, & i\%2 \neq 0 \\ \sum_{j=1,\ j\neq i,\ j\%2=0}^{M} X_j, & i\%2 = 0 \end{cases}, \tag{4.4}$$

all operations above are in binary field, and % denotes modulo 2 residual. Again, if user N_i does not have all packets required to generate $C'_{r(i)}$, then it will still be in selfish mode and send only its own packet.

When $M<6$, the transmission the stage three for the deterministic protocol needs to be modified as it will be shown in the special cases 4.3.1.9

4.3.1.6 Stage 3 2-Packet Deterministic Combination

To reduce the possible number of users in selfish mode that would occur in the above strategy, we propose another simple, deterministic odd-even strategy whereby only two packets are combined as shown in (4.5).

$$C'_{r(i)} = X_i \oplus X_{i+2}. \tag{4.5}$$

Odd numbered users combine only the odd neighbour's received packet X_{i+2} to their own packet. The algorithm is symmetric for even-numbered users. Again addressing is circular as in subsection 4.3.1.2.

4.3.1.7 2-Packet Non-Deterministic Combination

This strategy is a fusion of the strategies proposed in subsections 4.4.2.3 and Section 4.4.2.4. Odd-numbered user N_i combines its own packet with any other odd-numbered N_j's packet, such that $|i-j|$ is the minimum from all possible received X_j's at N_i. The scheme is symmetric for even-numbered users and boils down to the strategy of subsection 4.3.1.6 if $|i-j|=2$. A 1 bit header is needed to inform D of j.

4.3.1.8 *Stage 3 All-Received Non-Deterministic Combination*

This strategy is the same for 4.3.1.4 where each user combines its own packet to all received packets before forwarding the resulting packet to the other users and destination. *M* bits header is needed.

4.3.1.9 *Special Cases:*

When M<6, the transmission stages will be as shown in Table 4.1 which has been arranged to give M rank received matrix at D with any M+1 unique packets at the end of stage 2. Moreover, as always stage 3 combines two packets, we used two packets deterministic protocol for the third stage when M=5 and M=4, and simply added everything when M=3 and M=2. This way of combination was chosen to satisfy the condition that M+1 unique equation is always enough at the end of stage 2.

Table 4.1: Transmitted combined packets at second and third stage when M<6.

	M=5		M=4		M=3		M=2	
N_i	St2	St3	St2	St3	St2	St3	St2	St3
1	$X_2 \oplus X_3 \oplus X_4 \oplus X_5$	$X_1 \oplus X_2$	$X_2 \oplus X_3 \oplus X_4$	$X_1 \oplus X_2$	$X_2 \oplus X_3$	$X_1 \oplus X_2 \oplus X_3$	X_2	$X_1 \oplus X_2$
2	$X_1 \oplus X_3 \oplus X_4 \oplus X_5$	$X_2 \oplus X_3$	$X_1 \oplus X_3 \oplus X_4$	$X_2 \oplus X_3$	$X_1 \oplus X_3$	$X_1 \oplus X_2 \oplus X_3$	X_1	$X_1 \oplus X_2$
3	$X_1 \oplus X_2 \oplus X_4 \oplus X_5$	$X_3 \oplus X_4$	$X_1 \oplus X_2 \oplus X_4$	$X_3 \oplus X_4$	$X_1 \oplus X_2$	$X_1 \oplus X_2 \oplus X_3$		
4	$X_1 \oplus X_2 \oplus X_3 \oplus X_5$	$X_4 \oplus X_5$	$X_1 \oplus X_2 \oplus X_3$	$X_1 \oplus X_4$				
5	$X_1 \oplus X_2 \oplus X_3 \oplus X_4$	$X_5 \oplus X_1$						

4.3.2 Probability Analysis

In this section, we determine the overall probability of destination D being able to recover all M source messages given that it has received less than 2M packets during the first two stages from the M node. We assume that all nodes and the destination are within transmission range of one another. Moreover, all channels are assumed to behave in the same way.

After the first transmission stage, the destination will receive k packets from the M nodes, where $k \leq M$ different packets have been received by D at the end of stage 1 with probability:

$$P_1(k) = \binom{M}{k}(1-p)^k p^{M-k}. \tag{4.6}$$

The probability that the decoding will be complete after the first stage is simply:

$$P_{d,1} = P_1(M) = (1-p)^M . \tag{4.7}$$

If decoding is not finished after the first stage, the second stage of transmission will follow.

4.3.2.1 Benchmark PEP for Stage 2

We are seeking the Packet Error Probability (PEP) to finish the decoding after stage 2 for the benchmark protocol where each user sends its own packet twice.

So, the PEP for the packet to not be received after stage 2 is p^2, so, the PEP to be received is: $(1-p^2)$, accordingly, *D* will finish decoding all *M* packets after the second stage as shown in (4.8):

$$PER = (1-p^2)^M . \tag{4.8}$$

Accordingly, the probability of receiving *k* packet after stage 2 is shown in (4.9):

$$P(k) = \binom{M}{k}(1-p^2)^M .p^{2(M-k)}.\frac{k}{M}. \tag{4.9}$$

4.3.2.2 PEP for M-1 Deterministic combination stage 2:

In this stage, each user will either switch to fully cooperative behaviour (if it has received and decoded all the $M-1$ partners' packets) with probability $P_c = (1-p)^{M-1}$, or the user will remain in selfish mode with probability $P_S = (1-P_c)$.

After the second stage of transmission is finished, we are interested in the probability $P_{d,2}$ that the decoding at the destination is successfully completed. This is given by:

$$P_{d,2} = \sum_{k=0}^{M} P_{d,2}(k)P_1(k), \tag{4.10}$$

where $P_{d,2}(k)$ is the conditional probability that the decoding will be successful after the second stage, given that the destination received k user packets in the first stage from all users.

As we have *M* different packets, *D* must receive *M* linearly independent equations, i.e. any number of different equations that gives rank *M* receiving matrix at *D*. Accordingly, *D* evaluates the received rank matrix, and in the case of having *M* rank, *D* informs the *M* users that all packets have been recovered; otherwise, more transmission's stage is needed or *D* will declare that one or more uses are dead.

As the combination in the second stage is deterministic, the *M* users generates maximum of 2*M* unique packets. However, *D* needs to receive just *M*+1 unique packet to guarantee obtaining rank M receiving matrix.

In the case when *D* cannot hear from two users, or the case of dead users, *D* cannot receive *M* linear independent packets, which means that the received matrix's rank is less than *M*, though *M*+1 unique linear equations have been received.

Accordingly, we can confirm that *D* successfully decodes all *M* users' packet when *M*+1 unique packets received at *D* gives rank *M* received matrix at *D* as shown in Figure 4.2 (a) and Figure 4.2 (b), and (4.11) and (4.12) show the PEP for these cases. The decoding will successfully finish if for all $(M+1) - k$ time slots in which transmission failed in the first time slot, the transmission is successful in the second time slot. The decoding will succeed irrespective of the behaviour (selfish or cooperative as long *M* rank linear equations received) of users in these $M - k$ time slots, this is illustrated in Figure 4.2 (a), where 1 denotes successfully received packet, 0 denotes lost packet, S/C denotes a packet received in either selfish or cooperative mode, and *X* denotes any possibility of either received or lost packet. However, *D* can guarantee to obtain the *M* rank receiving matrix when *D* receives the packet in cooperative mode for one user received in stage 1 as shown in Figure 4.1. This means that any *M*+1 different packets give *M* linearly independent equations, taking into consideration that we are seeking the definite probability to obtain the rank *M* received matrix at *D*, (4.11) shows the PEP for case 1 (a):

$$PEP_{case\ 1} = (1-p)^{M-k}.(1-(1-(1-p)Pc)^{k}). \qquad (4.11)$$

Moreover, even if one of M-k users' packets is not received in the first time slots, remains not received in the second time slots, the decoding will succeed if, in the remaining k time slots, the destination receives at least two user's packets in cooperative mode.

An example of this case is illustrated in Figure 4.2 (b) where C_r denotes a packet received in cooperative mode.

St1	**1**	**1**	**0**	**0**	**0**	**1**	**0**	**0**	**0**	**0**
St2	**c**	**x**	**s/c**	**s/c**	**X**	**X**	**s/c**	**s/c**	**x**	**s/c**

(a) Case 1: cooperative protocol

St1	**1**	**1**	**0**	**0**	**0**	**1**	**0**	**0**	**0**	**0**
St2	**c**	**c**	**s/c**	**s/c**	**X**	**X**	**0**	**s/c**	**x**	**s/c**

(b) Case 2: cooperative protocol

Figure 4.2 : M+1 received packets at D, a and b are the solvable cases.

So, for case 2 PEP is given by (4.12):

$$PEP_{Case\,2} = (M-k)p(1-p)^{M-k-1}.\left[(1-(1-p)Pc)^{k} - k(1-p)Pc(1-(1-p)Pc)^{k-1}\right]. \qquad (4.12)$$

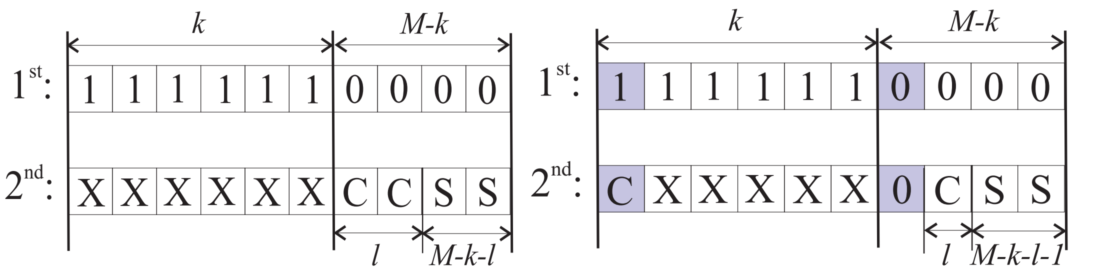
With these two successful decoding outcomes, we obtain (4.13):

$$P_{d,2}(k) = (1-(1-(1-p)Pc)^{k}\left[(1-p)^{M-k} + (M-k)p(1-p)^{M-k-1}\left(1-\frac{k(1-p)Pc}{(1-(1-p)Pc)^{k-1}}\right)\right], \qquad (4.13)$$

where P_C is the probability to receive the packet in cooperative mode.

Though M+1 unique equations always give M linear independent equations, which is enough to recover the M received packets, D can obtain the M linear independent equations from just M packets as shown in the following cases:

The probability $P_{d,2}(k)$ can be calculated by analyzing the probability of the two scenarios that lead to successful decoding of all M nodes after stage 2, illustrated in Figure 4.3.



(a) Solvable Case 1. (b) Solvable Case 2.

Figure 4.3: M received packets at D, a and b are the solvable cases.

Before describing them, we note the following important fact. Let $l \leq M$ be the number of combined equations received from l different nodes. We focus on the $l \times l$ subsystem containing these l combined equations, restricted only on the corresponding l nodes. This subsystem has full rank l if $l\%2 = 0$, i. e., the even number of combined packets for subsystem of l has full rank received matrix, hence, its solvability.

Figure 4.3 (a) illustrates the first solvable case when, after receiving $k \geq 0$ selfish equations in the first stage, in the second stage, each of the remaining $M-k$ nodes succeed in transmitting either selfish or combined equation. If out of these $M-k$ equations, $l \leq M-k$ are combined equations, the system is solvable, otherwise, it is not solvable. Note that whatever is received by k nodes already received in the first stage is irrelevant. The probability of the first solvable case is:

$$P_{d,2}^{(1)}(k) = \sum_{\substack{0 \leq l \leq M-k \\ l\%2=0}} \binom{M-k}{l} P_C^{\,l} P_S^{\,M-k-l}. \tag{4.14}$$

Figure 4.3 (b) illustrates the second solvable case when, after receiving $k \geq 1$ selfish equations in the first stage, in the second stage, $M-k-1$ out of the remaining $M-k$ nodes succeed in transmitting either selfish or combined equation, and one node is not successful for the second time. Then, if during the second stage at least one combined equation is received from any of the k nodes already received in the first stage, the system is solvable. Note that in this case, the number $l \leq M-k-1$ of combined equations among the $M-k-1$ combined schemes for different M's equations received by nodes unsuccessful in the first stage but successful in the second stage, is irrelevant. The probability that the second solvable case occurs is:

$$P_{d,2}^{(2)}(k)=(1-(1-(1-P)P_C)^k)\cdot(M-k)(1-P)\cdot\sum_{\iota=0}^{M-k-1}\binom{M-k-1}{\iota}P_C^{\iota}P_S^{M-k-\iota-1}. \tag{4.15}$$

If two or more nodes do not succeed in transmitting any equation to the destination node during both stages, the system cannot be solved. Therefore, the probability $P_{d,2}(k)$ for the two-stage $M-1$ deterministic combined system is:

$$P_{d,2}(k)=P_{d,2}^{(1)}(k)+P_{d,2}^{(2)}(k). \tag{4.16}$$

Replacing (4.16) in (4.10), the probability $P_{d,2}$ is obtained.

If D fails to receive at least M linearly independent equations from M, a third transmission stage is considered whereby novel information in the form of linear combinations not previously received at D is transmitted. We propose deterministic and non-deterministic combination strategies for stage 3 next. This stage also requires M time slots for transmission. D can recover the M nodes' messages when M linearly independent equations are received from all three stages.

Then, probability Pd,3 of successful decoding after three stages can be calculated similarly as in (4.16), as D will declare the ability to decode all M packets when receiving rank M matrix. Probabilistic analysis for other schemes is possible but in some cases tedious. We leave the complete probabilistic analysis for our future work, and explore the performances of all the schemes, including the three stage transmission, using simulations in the following section. Figure 4.4 shows the block diagram for the system design with NC over the erasure channel.

Figure 4.4 shows that the users tend to check the packets reception from the first stage transmission, and if not received, it tends to retrieve the packet by the helps of the combined packets received by the second cooperative stage.

If any packet neither received nor retrieved, is then asked to be retransmitted unlike the scenario where NC is not applied which is retransmitting the packet in the case when it is not received only.

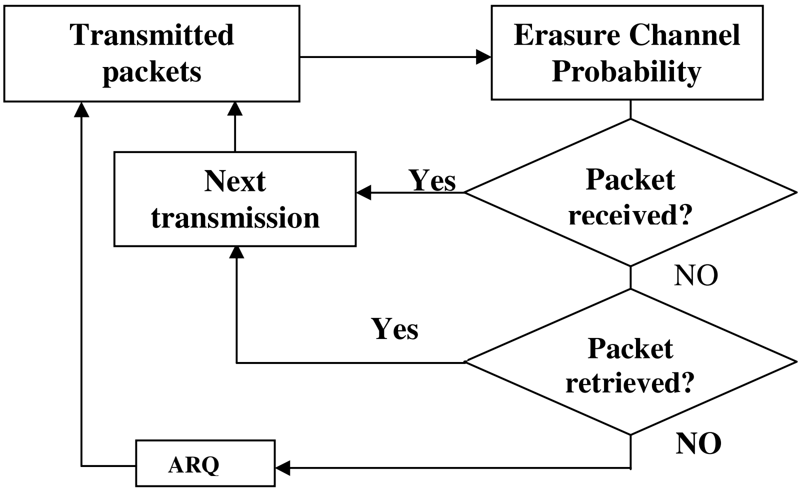


Figure 4.4: System set-up Block diagram with NC, the figure justifies the improvement in PEP.

4.4 LTE-A Proposed System for Downlink Scenario

Recently, a lot of attention has been devoted to transmission over LTE-A application, benefiting from Release 10 that provides HARQ and CoMP features that decreases the ARQ and supports cooperation over the network. CoMP is being applied over the physical layer; however, we are proposing applying it over the MAC layer believing that this enables the network to be more tolerant to drop packets with no need to retransmission.

In this Section, we apply NC in the deterministic way we mentioned in Section 4.3 over LTE-A downlink scenario, were a source of data sends *k* stream of packets to a Pico relay to forward to Femto relay(s) and users as shown in the following Sections.

4.4.1 LTE-A System Base Stations/Relays

The proposed scenario under investigation consists of three of relays intend to forward *k* packets of data streams to one user over LTE-A network as shown in Figure 4.5.

We assume that the relays under considerations are the ones that are in range with the Pico relay or source and with the user, i. e. any relay does not receive the packets and it is not able to forward them to the user; this relay is not included in our scenario.

We force our system model generic over the practical case of LTE-A which is the case when relays are connected via fiber optic connection with q=0 erasure probability, i. e. assuming that the two Femot relays have all all packets from the Pico relay.

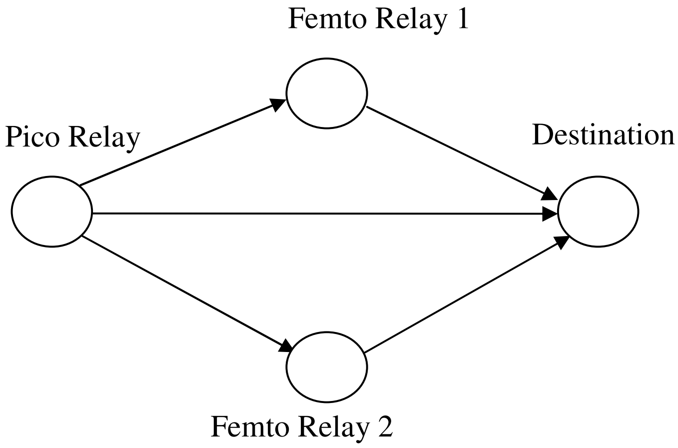


Figure 4.5: Pico relay connected to two Femto relays and one user as an example.

So, we have three erasure probabilities, which are, from the Pico relay to the user and from the two Femto relays to the user as shown in the Figure 4.5.

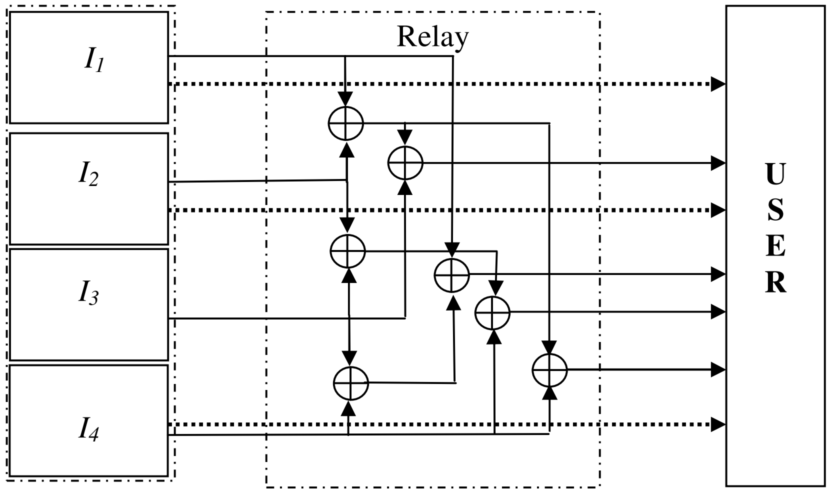
4.4.2 Deterministic Network Coding

Instead of just forwarding the packets from the two Femto relays to the user separately, NC could be applied in deterministic ways to improve the communication and save the data rate. Indeed, our analytical results show that applying NC results to decreasing the need for ARQ. The reason for this is the gained ability for the user to retrieve the unreceived packet rather than asking for the repetition.

There are different ways that *k* packets can be determinately combined. The reason we tend to use deterministic combination is that we are searching for maximum diversity, i.e. not to broadcast the same packet more than once to obtain as many novel packets as possible, moreover, deterministic combination makes the headerfile as short as possible. Following, we show the proposed deterministic combinations.

4.4.2.1 K-1 Deterministic Combination Femto Relay

The Femto relay performs the combination mentioned in Section 4.3.1.1 and (4.1). Figure 4.6 shows the combination the Femto relays performs before forwarding the packets to the users, four packets division is assumed as an example.



: Direct un-combined from Pico BS

: combined through the Femto relay

Figure 4.6: Pico base station and Femto relay transmitted packets to users; *k*-1 deterministic combination is applied over the Femto relay.

Notice that the Femto relay combines *k*-1 packets together, resulting to *k* unique packets. So, the Femto relay broadcasts: $I_2+I_3+I_4$, $I_1+I_3+I_4$, $I_1+I_2+I_4$, and $I_1+I_2+I_3$, taking into consideration that the Pico BS directly broadcasts I_1, I_2, I_3, and I_4.

In such case, the transmitted combined packets from the BS are shown in (4.17) bellow:

$$\begin{bmatrix} CT_1 \\ CT_2 \\ CT_3 \\ . \\ . \\ CT_4 \end{bmatrix} = \begin{bmatrix} I_1 & I_2 & I_3 & I_4 & .... & I_M \end{bmatrix} \begin{bmatrix} 0 & 1 & 1 & 1 & .... & 1 \\ 1 & 0 & 1 & 1 & .... & 1 \\ 1 & 1 & 0 & 1 & .... & 1 \\ . & . & . & . & .... & . \\ . & . & . & . & .... & . \\ 1 & 1 & 1 & 1 & .... & 0 \end{bmatrix}, \tag{4.17}$$

Where CT_1 is the combined packet for all packets transmitted by the BS.

We name the uncombined packet sent by the Pico BS by the *single* packet, and the combined packet without this single packet by the *complementary*, for example, $I_2+I_3+I_4$ is the complementary packet for the single I_1. This definition has important meaning in the PEP analysis.

In such combination, the Femto relay adds just a log_2k+1 bits header to confirm which packet is transmitted and whether it is single or complementary.

The Femto relays that perform this deterministic combination are assumed to have all k packets as our scenario is for optical fiber connections between the Pico BS and the Femto relays.

Under such scenario, each user will be receiving $2k$ unique packets, in term of k packets without applying NC that received by the Pico BSs, and k unique combined packets from the Femto relay.

4.4.2.2 k/2 Deterministic Combination Femto Relay

In this protocol, the Femto relay linearly combines $k/2$ packets, resulting in a simple, deterministic solution as shown in Section 4.3.1.5 and (4.4), where odd numbered packets in each complementary combined together and even numbered packets in each complementary combined together, resulting to k unique packets.

As example, assuming $k= 6$, the Femto relay generates six combined packets, in term of three odd combinations and three even combinations as following:

I_3+I_5, I_1+I_5, and I_1+I_3 odd packets combinations where I_3+I_5, is the combined packets for odd packets in the complementary of single packet I_1 and I_4+I_6, I_2+I_6, and I_2+I_4 even combinations where I_4+I_6 is the combined packets for the even users in the complementary of the single packet I_2.

In such combination, the Femto relay adds just a log_2k bits header to confirm which packet complementary is used to perform odd or even combination which means what packets are combined.

When $k<6$, the deterministic combined packets protocol generated by the Femto relay is shown in Section 4.3.1.9 for both k-1 and odd-even $k/2$ deterministic combinations.

4.4.2.3 2-Packet Deterministic Combination Network Coding

To increase the number of unique generated packets by the Femto relay, we propose two packet deterministic combinations, whereby only two packets are combined.

In both, the Femto relay adds just a $2log_2k$ bits header to confirm which two packets are combined.

This way of combination is recommended when the channel suffers from high noise values with big error burst, in such case the Femto relay will need to receive just one packet to combine with each packet rather than receiving such big number of packets. However, we propose two deterministic two combination packets as an example, though any two packets combination could be used.

4.4.2.4 2- Odd-Even Strategy Packets Deterministic Combination Femto Relay

In this combination, the Femto relay generates *k* unique deterministically combined packets whereby only two packets are combined as shown in Section 4.3.1.6 and (4.5) so

$$C_{2-odd-even(i)} = I_i \oplus I_{i+2}, \tag{4.18}$$

Odd numbered packets combine only the odd neighbour's packet I_{i+2} to their own packet. The algorithm is symmetric for even-numbered packets. Finally, packets are addressed in a circular order. This way of combination is used when the FSMC does not suffer from such big burs error trend as it is less expected to not receive two packets in following to each others.

4.4.2.5 2-Packets Deterministic Combination Next Neighbour only Femto Relay

In this combination, the Femto relay generates k unique deterministically combined packets whereby only two packets are combined as in Section 4.3.1.2, the only different is that the Femto relay tempts to combine the direct next neighbour as shown in (4.19).

$$C_{2-next(i)} = I_i \oplus I_{i+1}, \tag{4.19}$$

Packets are addressed in a circular order as well.

Finally, we can generalize the two packet combination by the following equation:

$$C_{2-next(i)} = I_i \oplus I_{i+b}, \tag{4.20}$$

Where *b* could be any number, and the value of *b* depends on the error burst size, i. e. when the error burst size is big, *b* tends to be big to be sure that we are choosing a packet out of the burst error and vice versa.

To improve the reliability even more, the system could implement more than one Femto relay. Moreover, each Femto relay can perform different NC combination to obtain even more unique packets. However, results show that the same combination through two Femto relays could be better than different combinations at specific erasure probabilities.

Following, we propose the scenarios when one Pico BS with one Femto relay, two Femto relays with same deterministic combination, and with two different deterministic combinations, with PEP analysis.
We show the PEP analysis through the simulation results as the theoretical analysis is similar to the uplink data gathering scenario.

4.4.3 Benchmark Femto Relay Scenario

In this scenario, we assume that the Femto relay does not apply NC over the received videos streams. In such case, the Pico BS broadcast the packets directly to the user and to the Femto relay which forwards the received packets to the user via a different channel. In such case, the PEP for a packet to be received is: $(1-p_P p_F)$ where p_P and p_F are the Pico and Femto relays erasure probabilities respectively.
In the case of two Femto relays, the probability for a packet to be received is $(1-p_P p_{F1} p_{F2})$ where p_{F1} and p_{F2} are the erasure probabilities for the first and second Femto relays respectively.
Following, we are studying PEP for different numbers of Femto relays and different deterministic combinations and we keep comparing our results with the benchmark scenario.

4.5 Cooperative Network Coding Femto Relay(s) Scenario

In this scenario, the Femto relay performs NC over the packets received from the Pico BS rather than just forwarding them.
We are proposing that the Femto relay performs the NC in a deterministic way as shown above, and then results are compared with the similar benchmark scenario.
Simulation results are adapted in this scenario as the analytical way has been shown in the first scenario which is enough to show the way that we perform it.

4.5.1 Cooperative Pico BS and one Femto Relay *k*-1 Deterministic Combination

In this case, the Femto relay will perform NC over the *k* packets in the deterministic way shown in (4.1). The simulation analysis shows the performance for this case as the theoretical analysis has been done over the uplink communication scenario and it is similar to this downlink scenario.

As the combination in the Femto relay is deterministic, the Pico BS and Femto relay generate $2k$ unique packets, assuming that the Femto relay has all the k packets. However, each user needs to receive just k+1 unique packet to guarantee obtaining rank k receiving matrix.

As a result of applying this deterministic combination, each user can recover the k packets even if it does not receive both of one single packet and it's complementary, which is equal to the probability by not receiving the same packet from the Pico BS and the Femto relay in the case when NC is not applied. When applying NC, it is enough to receive k unique linearly independent equations to recover the k unknown packets and, as we are combining the packets at the Femto relay in a linear deterministic way, it is guaranteed that this un-received single packet will be in all of the combined packs excepts this singles' complementary packet, as shown in (4.1). So, this single packet is being received indirectly through all the other combined packets (complementary), which makes it possible to be retrieved Moreover, as in the uplink scenario, it is enough for the user to receive any k+1 out of the $2k$ unique packets to recover all k unknown packets as in this scenario, the maximum received $2k$ unique packets as the same packets proposed in the uplink two stages communication.

Similarly**,** it is impossible to recover the k packets at the user if two single packets and their two complementary are not received at the user, and the reason for this was shown in the uplink scenario which is if two single packets and their complementary have not been received, the rest of the packets received from Pico BS and Femto relay give ($2k$-4) give k-1 linearly independent equations, hence, the impossibility to recover all k packets. Second Femto relay or ARQ is needed to solve this problem.

The retrieving process starts with the received direct packets from Pico BS, and in the case of not received single packets, the user resorts to the complementary packets received from the Femto relay to retrieve the requested packet(s), with no need to request the re-transmission.

Accordingly, the number of retrieving steps depends on the number of packets received directly from the Pico BS, in term of the more packets received directly from the Pico BS, the fewer retrieving steps are needed. Moreover, the more retrieving steps are needed the more BER aggregation as shown in the results part when testing this combination over the physical layer.

So, applying NC over the Femto relay allows the system to be able to loss k-1 packets without the need for retransmissions as long as no more than one single packet and its complementary packet are not received, hence, this shows that applying NC over Femto relay allows the system to retrieve the unreceived packet from Pico BS by the cooperation with the combined packets received by the Femto relay.

As the analytical processing is similar to the first scenario, we confine our analysis in this scenario over the simulated results for the two cases under investigation though the analysis is possible but it will be a manipulation of the solvable cases in this scenario in the exact way performed over the uplink communication.

4.5.2 Cooperative Pico BS and Two Femto Relays *k*-1 Deterministic Combination

We propose deterministic second Femto relay which gives novel deterministic combinations. This added Femto relay requires k time slots for transmission. The user can recover the k packets when k linearly independent equations are received from the Pico BS and the two Femto relays.

Comparing the performance for the different combinations has been performed in Section 4.4 in good details though it is for a different scenario, but it is still the same combinations.

Moreover, adding more than one Femto relay to the system makes it totally tedious to analysis the probability of recovering all k packets, though it is possible. However, it is enough to show the simulated performance for full reception as in the first scenario.

In this case, the Femto relay will perform NC over the k packets in either the same deterministic way shown in Equation 4.1, or in a different deterministic way from the one proposed in the uplink two and three stages communication.

To give reasonable illustration for the added second Femto relay, we do the case of adding other *k*-1 deterministic combination Femto relay as the analysis is relatively reachable.

In this scenario, we add a second Femto relay to the system mentioned in Section 4.5.1. This added Femto relay helps the users to collect the *k*+1 unique packets successfully.

The user will confirm decoding all *k* packets when and only when receiving *k* linearly independent packets from the Pico BS and the two Femto relays, which means receiving *k* linearly independent packets from the 3*k* packets sent from the Pico BS and the two Femto relays.

However, though we have 3*k* transmitted packets, we need to understand that the two Femto relays send the same packets resulting to duplicating the complementary packets, i. e. each single packet has its complementary packet sent from the two Femto relays with two different erasure probabilities.

Though this is good in term of not having two single packets without their complementary which means resorting to ARQ, but using the same combination in both Femto relays results to redundant transmission and reception in the case of receiving the same complementary packet twice.

Though it is less possible to not receive two single packets and their complementary as each complementary packet is being sent twice through two different channels, it still is the same problem not solved in this combination, i. e., if two single packets and their complementary have not received from the Pico BS and two Femto relays, the user will demand the retransmission. So, the second Femto relay improved the probability to not have this problem but basically, it does not solve it.

Finally, we need to remind that all the complementary packets give k-1 unique linearly independent equations, which means we always need one single packet to be received from the Pico BS though in the case of receiving the *k* packets through the two Femto relays, i. e., the cooperation between the two Femto relays does not solve the problem without the help of Pico BS.

In conclusion, implementing a second Femto relay which performs the same *k*-1 deterministic combination improves the probability to not repeat packets. However, it does not solve the problems as the added Femto relay does repeat broadcasting the same complementary packets the first Femto relay broadcasts, accordingly each complementary packet has better probability to be received but it does not give any extra novel information to be used in solving the problems.

As in one Femto relay scenario, the retrieving process starts with the received direct packets from Pico BS, and in the case of not received single packets, the user resorts to the combined complementary packets received from the two Femto relays to retrieve the requested packet(s), with no need to request the re-transmission from Pico BS.

Accordingly, the number of retrieving steps still mainly depends on the number of packets received directly from the Pico BS.

So, implementing the second Femto relay allows the system to be able to loss more packets than one Femto relay as long as k+1 unique packets received from the 3k transmitted packs that gives rank k received matrix.

The probability of full reception is shown in the simulation results.

To solve the problems in the k+1 deterministic combination, we recommend that the second Femto relay should obtain novel information as proposed in the following scenarios.

4.5.3 PER for Pico BS with Two Femto Relays *k*-1 and *k*/2 Odd-Even Deterministic Combination

In this scenario we propose to replace the second deterministic k-1 combination used for the second Femto relay by other deterministic combination to have novel linearly combined packets. This is because we want to solve the problems mentioned in Section 4.5.2. The reason we propose this combination as the k odd-even linearly combined packets gives k linearly independent equations as the rank of the transmission matrix is k. This is such important advantage because it means that receiving just these packets at the users is enough to solve the problem. In such case, we can confirm that we have solved the problem of not receiving two single packets with their complementary packets when the two Femto relays were proposed with the same k-1 combination. The user can retrieve the k packets even if not receiving any single packet from the Pico BS.

So, the Pico BS and the two Femto relays broadcast $3k$ unique equations, and each user aims to receive k linearly independent equations from the $3k$ linearly combined equations. If more than two Femto relays are implemented, we recommend more different deterministic combination such as two packets deterministic combinations as this gives more unique packets and avoid repeating the same transmitted packets which prevents the redundancy in transmission or reception.

The probability analysis is totally complicated though it is possible. However, the simulation results compare all combined protocols with different values of erasure probabilities for all channels have been shown in similar combination but different scenario in Section 4.4.

4.6 Simulation Results

We show the results for the uplink data gathering scenario and for the downlink LTE-A scenario, this is to show that NC can be equally profitable when applying in either way.

4.6.1 Simulation Results over Erasure Channel for the Uplink Scenario

This section presents the results and observations of our simulation experiments to determine how our proposed cooperative protocols based on NC reduces the probability that all M nodes are not successfully decoded, P_e, with no extra transmitted packets compared to the benchmark protocol. Figure 4.7 shows the behaviour of P_e when applying all the proposed protocols as compared to the benchmark protocol for transmission in two stages. Results show how deterministic protocols are as competitive to the non-deterministic (random) with respect to P_e, but with the advantage of by passing header information for deterministic protocols and the easier Gaussian elimination decoding as the received packets are either single or combined in a deterministic way. Moreover, we can see that the M-1 deterministic combination outperforms all received combined at high error rate i. e., p=0.25, and this shows how the deterministic combination is stable and can stand dropping large number of packets.

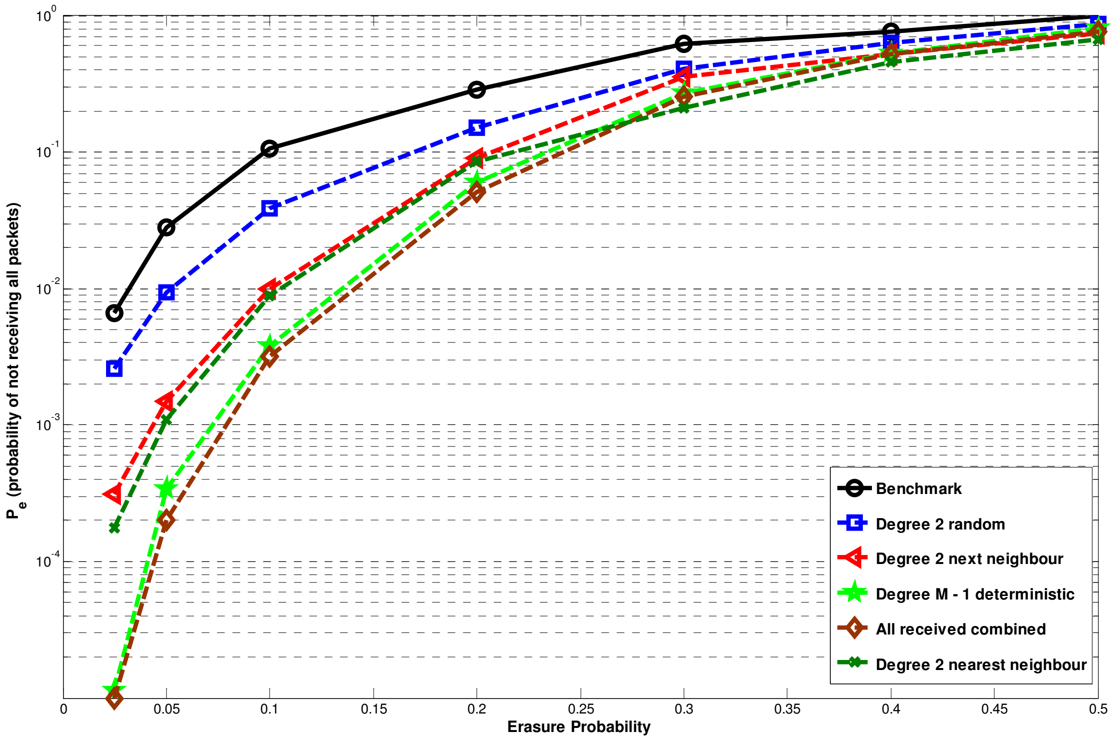


Figure 4.7: *Pe* for 2-stage schemes (M = 10 and q = 0).

Combinations of 2-packets provide acceptable P_e improvement though their performance is considerably weaker than both the $M-1$ deterministic and all received combined protocols. In addition, the deterministic degree 2 next neighbour deterministic protocol is as competitive as the closest neighbour protocol, but does not require header fields, this is justified by using the same erasure probabilities for all distances. The P_e behaviour of the $M-1$ deterministic combination protocol for different number M of nodes is presented in Figure 4.8, showing that the M-1 deterministic protocol is as competitive to the non-deterministic all-received combined protocol even for a large number of nodes. Figure 4.8 also shows that increasing the number of nodes M deteriorates performance of both cooperative protocols and the benchmark non-cooperative protocols.

Figure 4.8 shows the worsening performance of the $M-1$ deterministic combined protocol when increasing the erasure probability q between nodes. The threshold behaviour of q is notable, where the performance at low values of p is significantly improved if $q \leq 0.3$, and otherwise, deteriorate significantly.

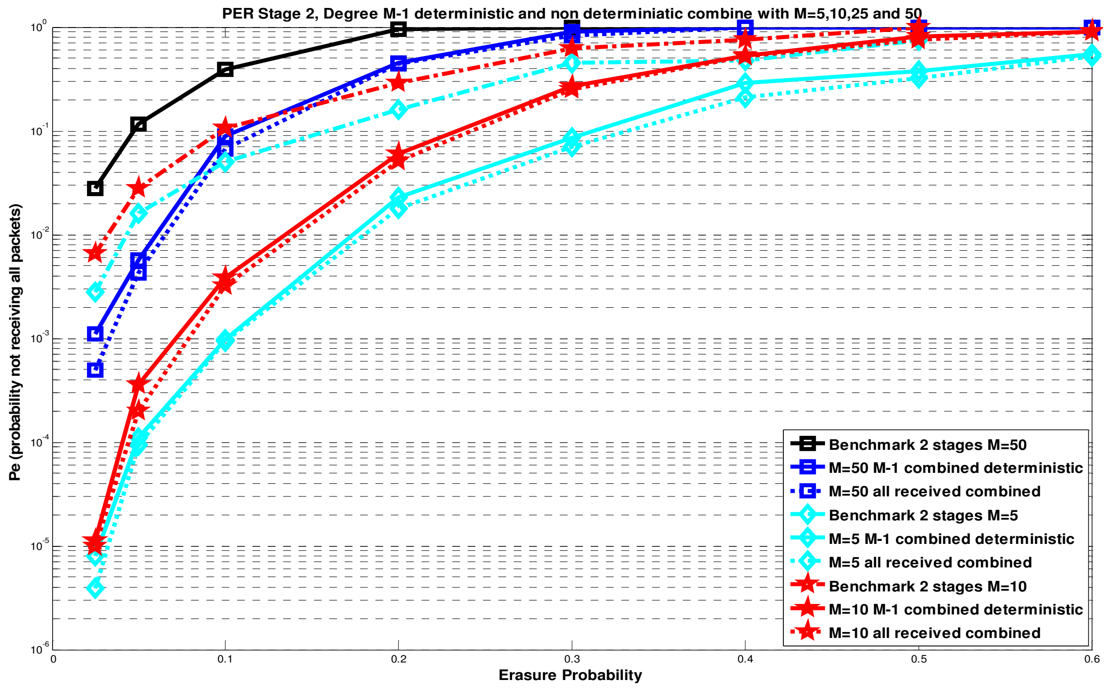


Figure 4.8: *Pe* for 2-stage *M*-1 deterministic and all received combined schemes for different *M*'s.

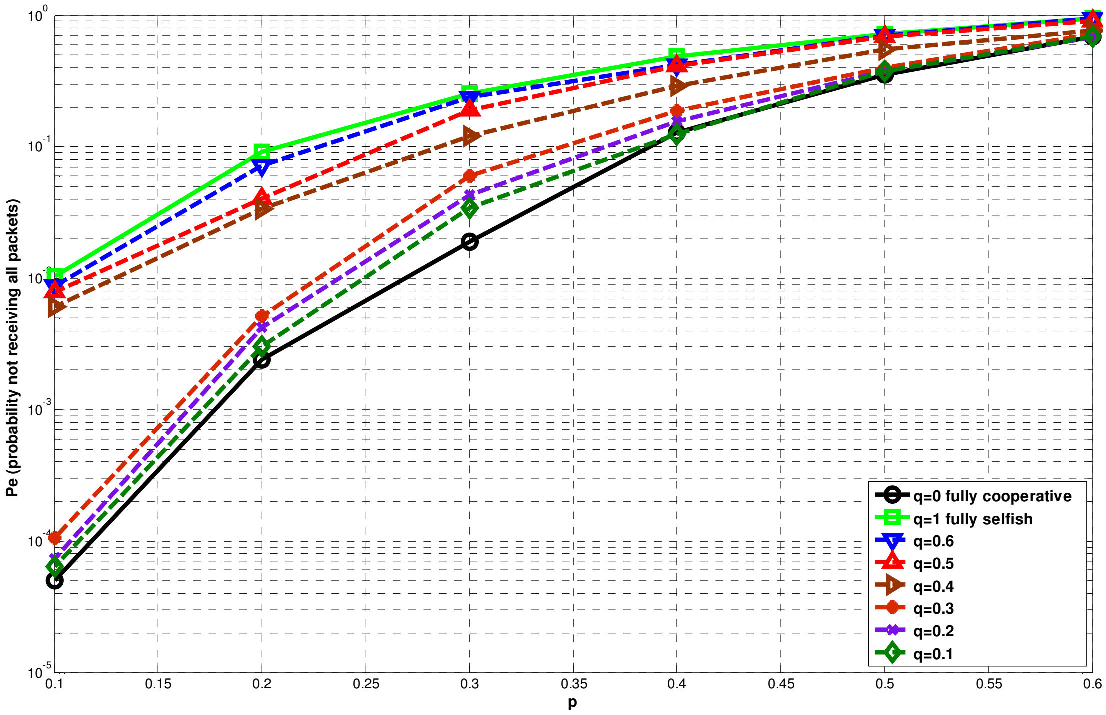


Figure 4.9: *Pe* for 3-stage *M*-1 deterministic combination example, $M = 10$ and different q's.

Figure 4.9 shows the M-1 deterministic combination when changing the erasure probability between the M users. The clear threshold in Figure 4.9 is justified by obtaining the sufficient number of unique linear equations, I.e., when more than 30% losses packets, it is becoming harder to obtain the unique M+1 linear equations.

In the final part of results, we focus on the deterministic three stages to give more details through the results as the theoretical analysis is tedious.

Figure 4.10 shows the performance of the deterministic system that transmits odd-even type (4.4) in the third stage, instead of repeating fully combined (4.1).

As noted in Sections 4.3.1.1, fully combined equations are used to recover from the case when one user was not received at all by the destination. The motivation for odd-even equations is that reception of these equations enables recovery of any two users that were unable to send data to the destination. However, the destination can make use of only one odd and even equation; the rest of possibly received equations of this type are useless. Note that the number of users that will send odd-even cooperative equation in the third stage depends on q.

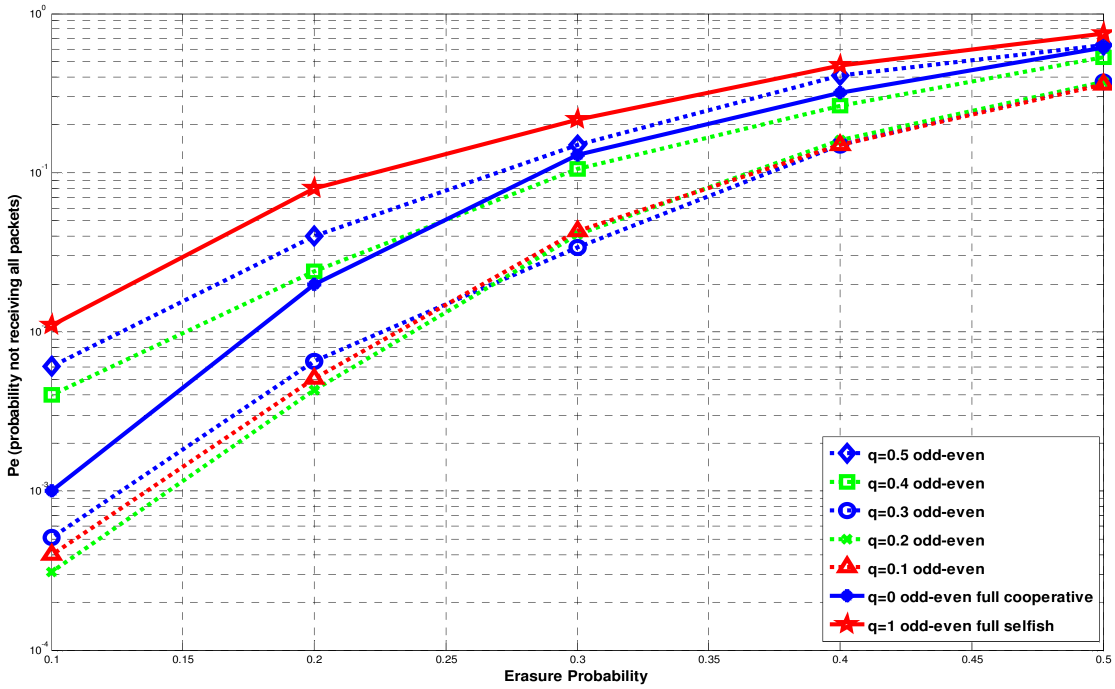


Figure 4.10: P_e performance of deterministic combined/odd-even cooperative scheme for three transmission stages.

This explains the PER results in Figure 4.10 where there is an optimum value of q (q = 0.2) for which the system performs better than for fully cooperative case q = 0.

Figure 4.11 illustrates the interval of q values, q = [0.3, 0.6] for which it is better to send odd-even (4.4) than to repeat the fully combined (4.1) equation in the third stage.

From the user viewpoint, the decision to send odd-even or fully combined equation during the third stage may be decided based on the previous channel state information. For smaller values of q, the strategies where a user randomly decides whether to send odd-even or fully combined equation with appropriate optimal probabilistic decision that depends on q have a potential to further improve the overall PER performance.

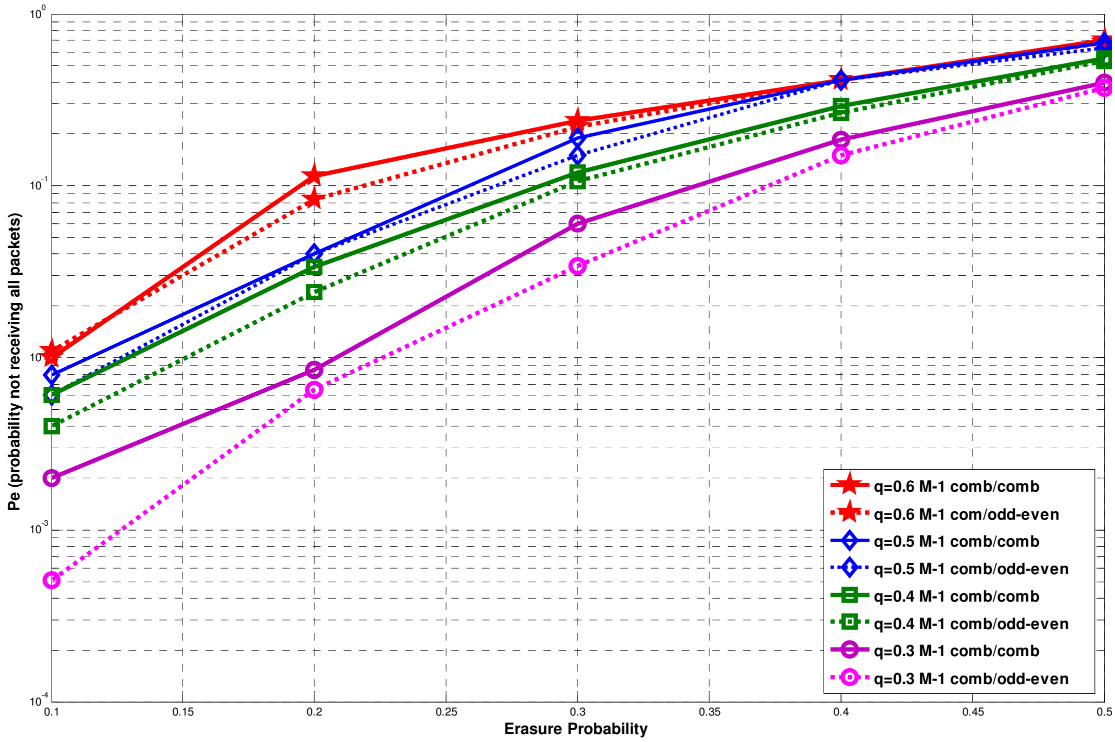


Figure 4.11: P_e performance of deterministic combined/odd-even cooperative scheme for three transmission stages.

Figure 4.12 shows the performance gained by the third transmission stage, when the channel between users are good (0.03), indeed, the P_e improved from 5.3 e^{-3} to $4e^{-5}$ at p=0.1, the cost of this good improvement is sending M extra packets.

When comparing three stages strategies, we notice that three stages deterministic out performs the three stages random combination clearly after the threshold p=0.3, which showed in Figure 4.8, though just one bit headerfile is required. We can notice that using random combination when p is more than 0.35 is better than our proposed deterministic combination, but it requires M headerfile.

Same observation is interestingly noticed at two stages, where our proposed deterministic out performs the two stages random combination just after p=0.3.

Moreover, Figure 4.12 shows that repeating odd-even in the third stage results to worsen performance than the random combination, so, it is suggested not to repeat odd-even combination in the third stage.

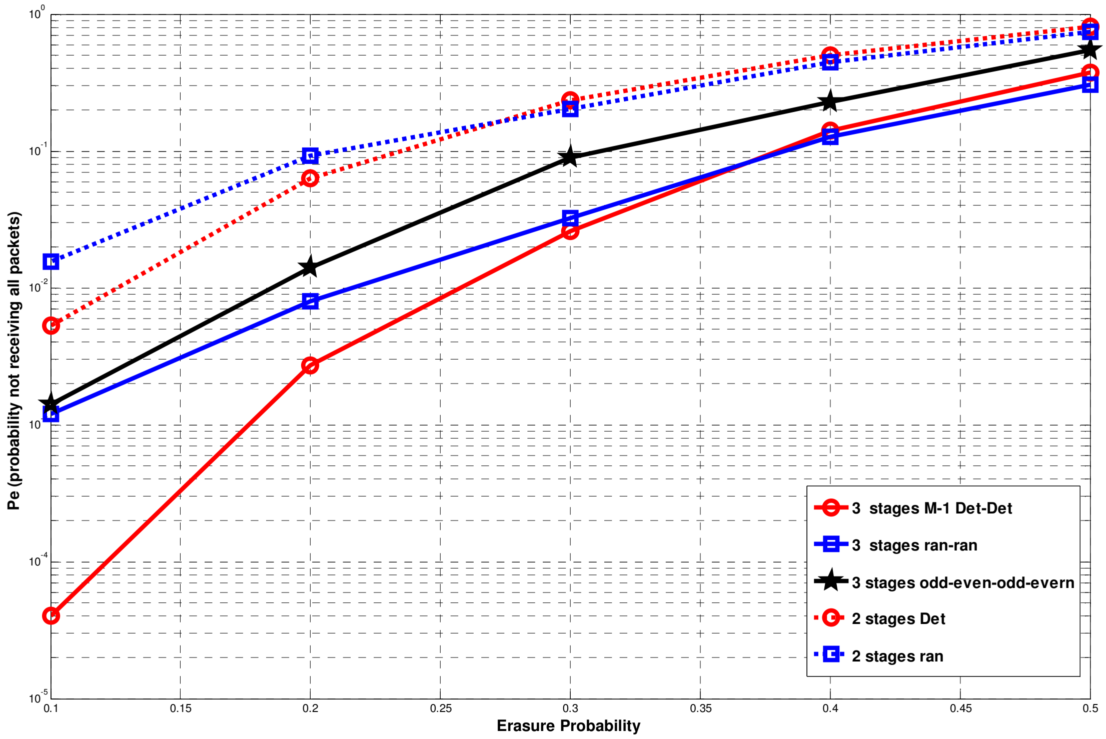


Figure 4.12: compare three and two stages for M=10 at good erasure probability (q=0.03).

4.6.2 Simulation Results for LTE-A Erasure Channel for the Downlink Scenario

The simulation results show the improvement in the network communication after applying NC over the Femto relays compared with the same scenario where NC is not applied.

We divide our results to Low Error Set (LES) and High Error Set (HES) at two different number of combined packets *k*.

In the case of LES, the Pico relay erasure probability is assumed to be 0.1, while it is as high as 0.3 in the case of HES, and we run the simulation for a small value of combined packets at *k*=10, and large value when *k*=50. In all running cases, $p_{F1} = p_{F2}$ to have fair comparison between the combination strategies.

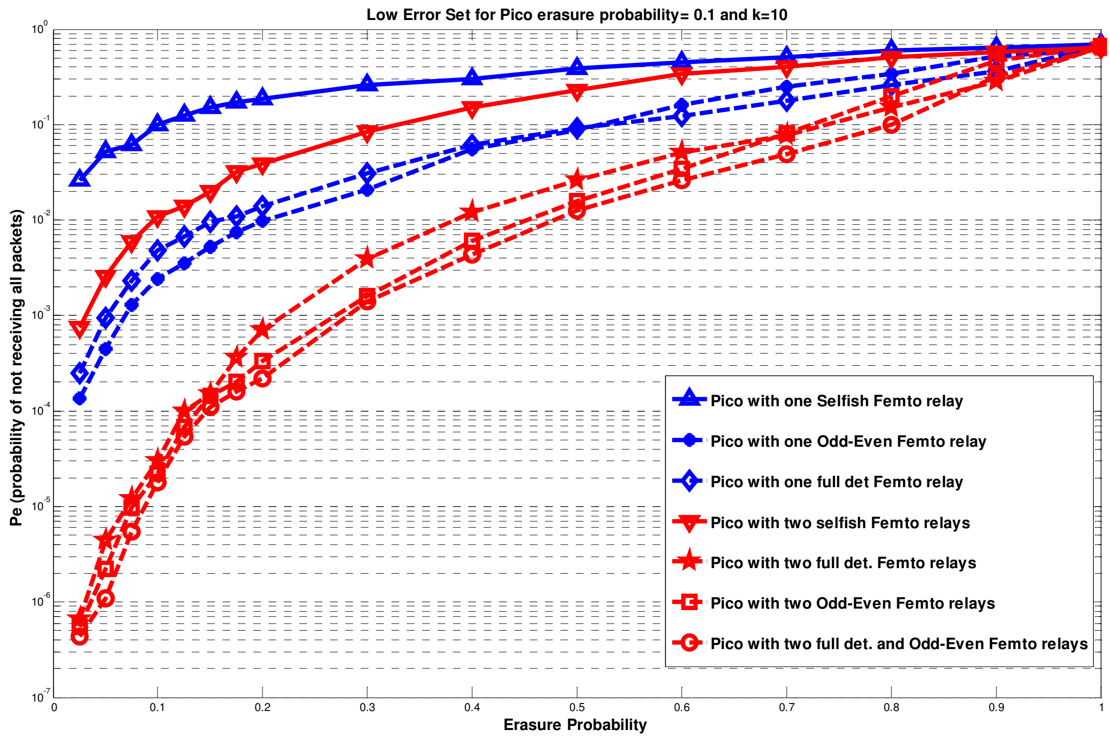


Figure 4.13: Low Error probability Set and small *K* =10.

Figure 4.13 shows probability of the full recovery in the down link for one user in the case of LES at k=10. Taking into consideration that all the improvement obtained by applying NC is measured here for only one user side, so, the improvement will be significantly maximized when the number of users increases.

Figure 4.13 shows the performance for one and two Femto relay in the case of benchmark and cooperative modes for LES and small combination value.

We can see how the cooperative mode outperforms the benchmark scenario where NC is not applied, indeed, at the erasure probability of 0.1 between the Femto and the user results to probability to not recover all the packets at the destination to be as low as 1.2 e^{-5} compared to 1.1e^{-2} when two Femto relays are implemented. Moreover, Figure 4.13 shows that both one and two Femto relays scenarios outperform their benchmark scenario significantly.

One more important observation from Figure 4.13 is that the best performance is obtained when the two Femto relays used two different combination strategy as this gives the best probability, in our case, one Femto uses full deterministic combination shown in Section 4.3.1.1 and the second Femto relay uses the Odd-Even combination shown in Section. 4.3.1.5, this is justified by the fact that the two different combinations provides more unique linear combinations which is the same results we obtained in the uplink communication. However, in the case of one Femto relay, we notice that the fully deterministic outperforms the Odd-Even till the erasure probability of 0.5 but after this point, Odd-Even combination outs performs the fully deterministic, and it is the same for two Femto relays scenario, where the fully deterministic two Femto relays outperform the two different combinations Femto relay till the point of 0.7 erasure probability, and then the fully deterministic and the Odd-Even combination Femto relays outperforms the two fully deterministic relays. The reason for this is justified by the way the deterministic combination is performed and with the probability to not receive two single packets with their complimentary.

Figure 4.14 shows that when increasing the number of combined packets, the performance decreases for all the scenarios, which is justified by the fact that more unknowns packets needs to be recovered. However, even in such large value like k=50, our proposed protocols well outperform the benchmark and give such acceptable probability.

In the case of HES where Pico erasure channel has 0.3 probability as shown in Figure 4.15, where *k*=50. We can see clearly the need for using NC cooperative as both benchmark scenarios totally decayed even in the case of two benchmark Femto relays. Figure 4.15 shows how applying NC over the two Femto relays revived the probability of 0.29 in the case of benchmark to 1.8e^{-4} when NC is applied over the two Femto relays. The Figure 4.15 shows again that using different combinations at the two relays gives better performance.

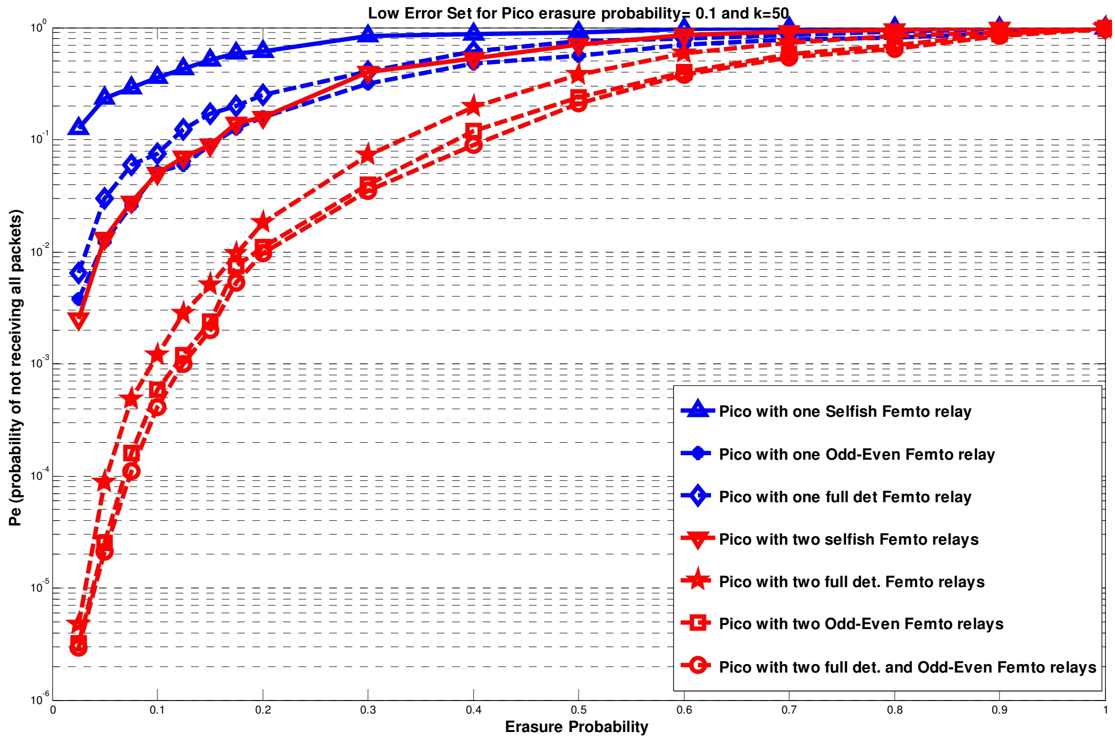


Figure 4.14: Low Error Set probability for large number of combination (K=50).

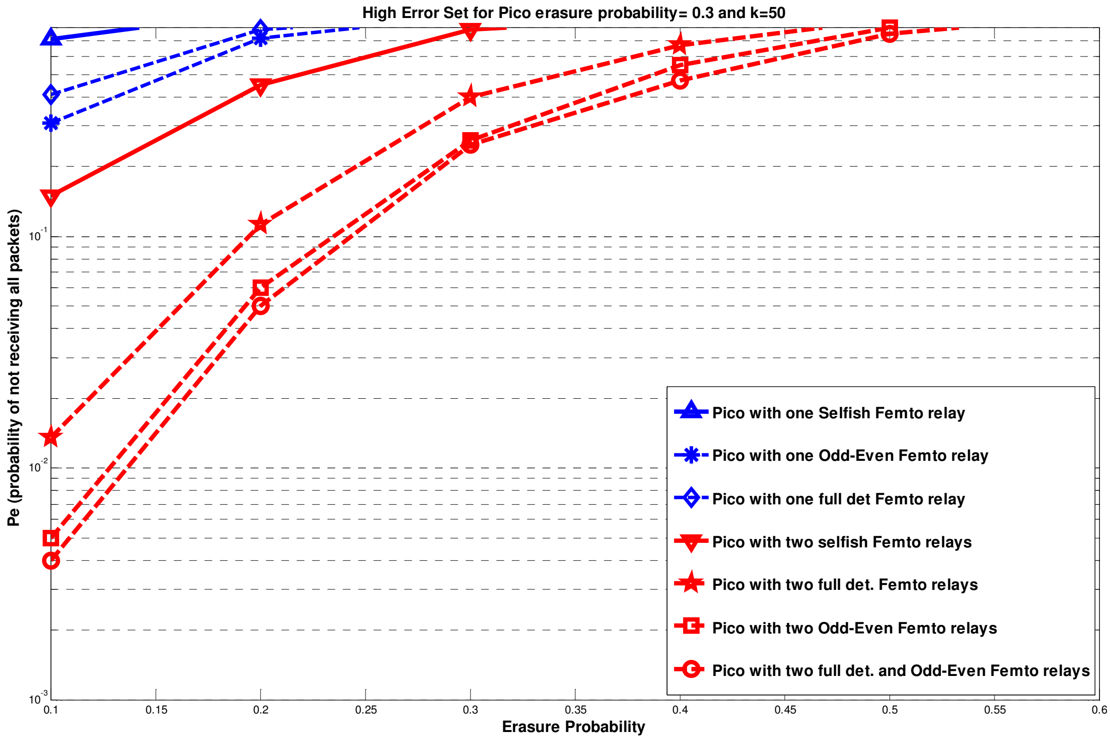


Figure 4.15: High Error Set probability and large K=50.

Figure 4.16 shows the fact that decreasing the number of the combined packets *k* from 50 to 10 results to such noticeable improvement in the probabilities and again even in small number of *k*, two different combinations gives the best results.

After comparing all the results, we come to Figure 4.17 which shows a full comparison between the scenarios (benchmark and cooperative) for LES and HES for the best combination found which is two different combinations over the two Femto relays.

Figure 4.17 shows that applying NC cooperative over the Femto relays results to such significant improvement for all the scenarios and under either error set probabilities and for any combined number of packets k.

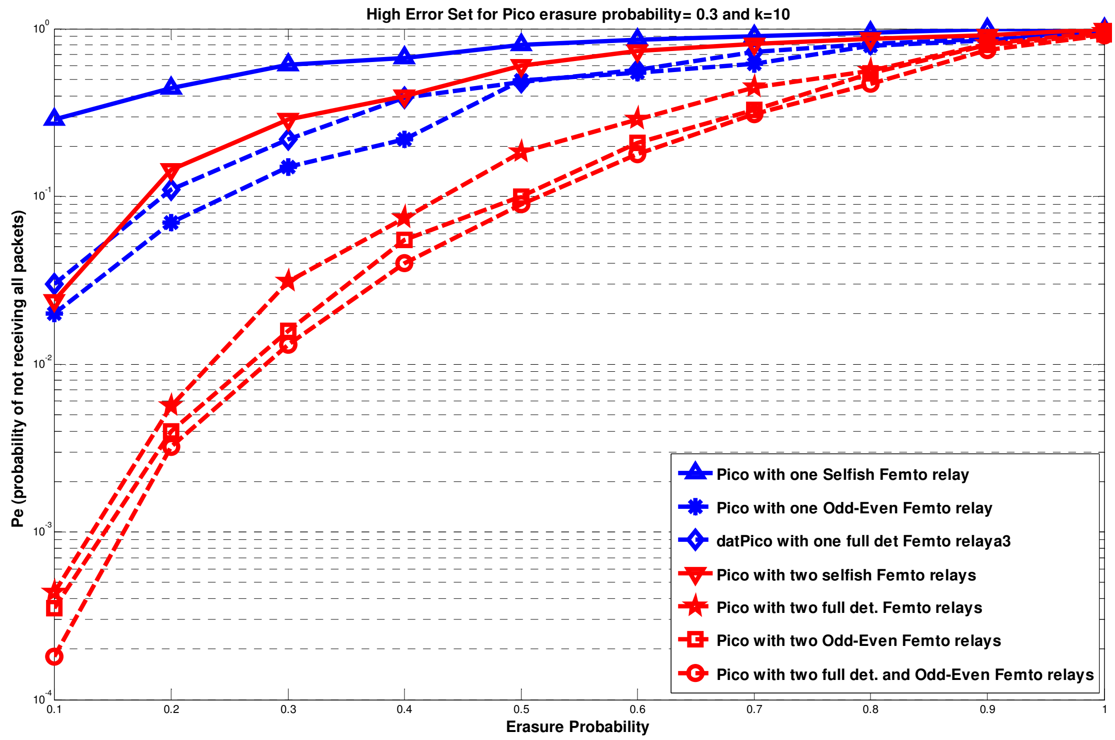


Figure 4.16: High Error Set probability for small number of combinations K=10.

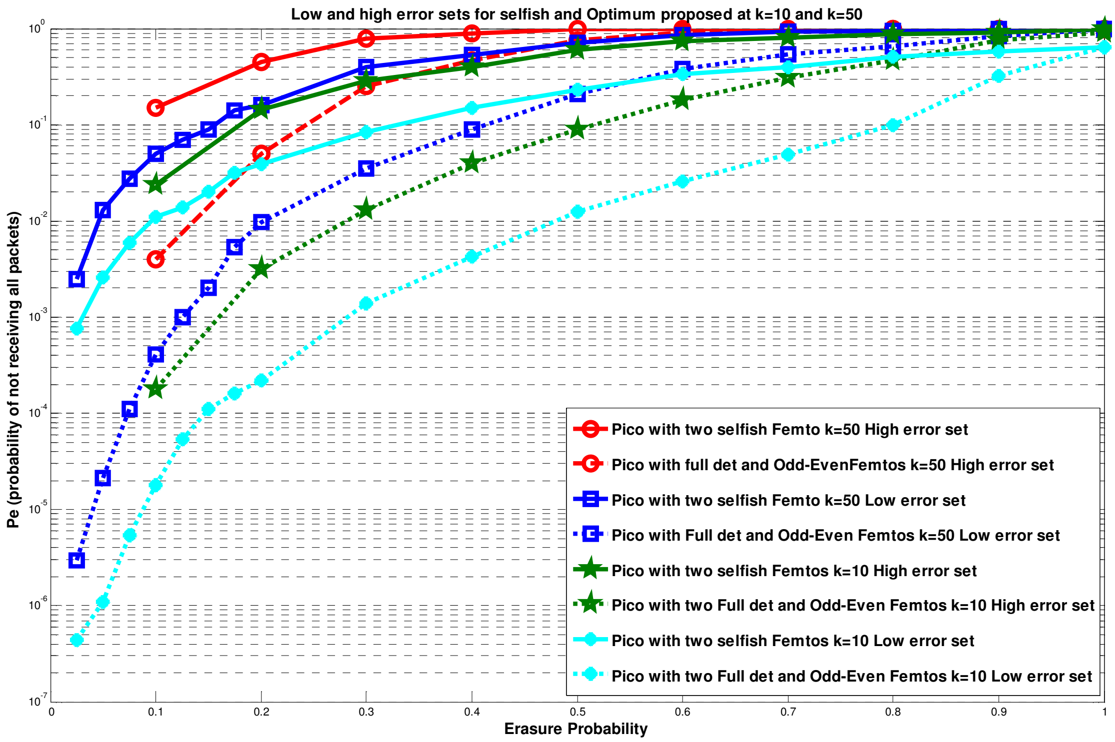


Figure 4.17: Low and high error sets for benchmark and Optimum proposed at k=10 and k=50.

Finally, we need to confirm that all results demonstrate the performance gain obtained by applying NC cooperation as compared to the benchmark case, taking into consideration that the proposed protocols use simple XOR combinations which are easy to implement and simple to decode at the user by using Gaussian Elimination.

4.7 Uplink and Downlink Network Coding over Erasure Channel Network Conclusion

Motivated by recent results from [106], [107] and [108] on the diversity and coding gains obtained by using Network Coding (NC) in cooperation protocols, we propose several practical transmission protocols for two different scenarios, a typical uplink localized data gathering scenario within a Wireless Sensor Network (WSN) cluster and over Long Term Evaluation Advanced (LTE-A) downlink three relays network scenario. Both scenarios show simplicity and low complexity while still maintaining performance gains over non-network coding solutions.

We discuss analytical and simulation results for the probability in the uplink scenario that the destination does not recover all nodes' packets, after two and three transmission stages adopting our proposed cooperative protocols based on low-complexity binary field NC. Simulation results demonstrate a perfect match with the analytic results and demonstrate the performance benefits of the proposed protocols over the baseline repetition protocol. In the downlink LTE-A scenario, we adapted the simulation results to illustrate our analysis and to present the improvement gained by applying NC over the Femto relays.

Moreover, we point out that the deterministic combining protocols are as competitive as the non-deterministic, but without the header overhead.

Finally, we showed that full connectivity in both the uplink and downlink scenarios are maintained even when the network drops a significant number of packets through the erasure channel, and we showed that this is justified by the ability of the sink to retrieve the propped packets than requesting them to be retransmitted. And just simple low-power Gaussian Elimination decoding process is needed to do so.

5. FSMC Erasure Channel Modelling for Uplink and Downlink Scenarios

5.1 Introduction

In Section 5.2, we introduce the FSMC technology with the theoretical limits and equations. This section is followed with Section 5.3, which shows the uplink and downlink Rayleigh Block-fating channel scenarios which are used in this chapter. Section 5.4 illustrates the simulation set-up and results for the Rayleigh fading channel to end the uplink transmission part in this chapter.

Section 5.5 shows the downlink scenario, represented by LTE-A network over FSMC with its introduction, and results in the subsection for the Section of 5.5. Finally, section 5.6 concludes the chapter.

5.2 Finite State Markov Chain

For realistic study, each wireless Rayleigh block-fading channel is represented by a packet-level Finite State Markov Chain (FSMC) channel model [110].

Following for our designed FSMC which has been introduced in Section 2.5.4 FSMC is used to represent the Rayleigh fading channel, which enables us to represent the relationship between the physical layer and its FSMC model, for a packet transmission system in the MAC layer. Thus, the channel is modeled with erasures or packet losses.

The received SNR values are partitioned into a K finite number of state-spaces, according to the channel fading speed $(f_D T_P)$,where f_D is the maximum Doppler Frequency and T_P is one packet time period.

The kth state-space duration is required to be some multiple of T_P, so, the duration of each signal segment τ_i is given by (5.1)

$$\tau_i = C_k T_P, \tag{5.1}$$

where k=1,2,…,K, and C_k is a constant criterion which depends on each state-space's duration, the state-space duration should be large enough to be more than one packet length, i.e. C_k is larger than 1, to guarantee that we are working under Markov chain principle, where transmission happens just between the adjacent states, as shown in (5.2)

$$P_{k,i} = 0 \text{ if } |k - i| > 1, \tag{5.2}$$

where $P_{k,i}$ is the packet transition probability between states k and i.

On the other hand, C_k should not be chosen to represent a too large number of packets in each state-space to maintain the same BER for all packets in the state. We assume using the same C_k all received SNR values.

The constant criterion C_k can be calculated from (5.3):

$$C_k = \frac{\exp\left(-\frac{\Gamma_k}{\gamma_0}\right) - \exp\left(-\frac{\Gamma_{k+1}}{\gamma_0}\right)}{\left(\sqrt{\frac{2\pi\Gamma_k}{\gamma_0}}\exp\left(-\frac{\Gamma_k}{\gamma_0}\right) + \sqrt{\frac{2\pi\Gamma_{k+1}}{\gamma_0}}\exp\left(-\frac{\Gamma_{k+1}}{\gamma_0}\right)\right) \cdot f_m T_P}, \tag{5.3}$$

where Γ_i is SNR partition level for state-space k and γ_0 is the average SNR.

To make the channels behaviour more realistic in reflecting the changing in the received SNR values, a dividing function is used to give each received SNR value the proper number of Markov chain state-spaces and the proper value of the partition's SNR threshold Γ_k, which depends on the received SNR value, as the constant criterion C_k and P_T are assumed to be the same for all received SNR values.

Determining the number of state-spaces K, and the threshold SNR partition Γ_k Γ_i for each state-space k=1,2…, K, is in increasing order where Γ_0=0, and Γ_{K+1}=∞, as shown in Figure 5.1.

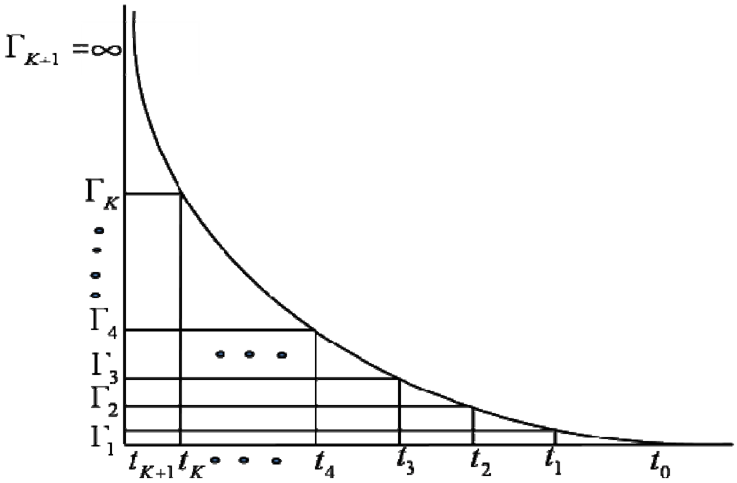


Figure 5.1: SNR partition values versus state-space time, $\Gamma_0 = 0$ and $\Gamma_{K+1} = \infty$.

The rest of Markov chain parameters are calculated from K and Γ_K, k=1,2,…K, starting from the steady state probabilities π_i as in (5.4):

$$\pi_k = \int_{\Gamma_k}^{\Gamma_{k+1}} p(\gamma)\ d\gamma = \exp\left(-\frac{\Gamma_k}{\gamma_0}\right) - \exp\left(-\frac{\Gamma_{k+1}}{\gamma_0}\right), \tag{5.4}$$

where $\sum_{k=1}^{K} \pi_k = 1$ and $p(\gamma)$ is the probability density function and given by (5.5)

$$p(\gamma) = \frac{1}{\gamma_0}\exp\left(-\frac{\gamma}{\gamma_0}\right), \qquad \gamma \geq 0. \tag{5.5}$$

Equation (5.5) shows that the received instantaneous SNR γ is distributed exponentially with the probability density function, as a result of the received signal envelope which has the Rayleigh distribution with additive Gaussian noise.

The steady state probabilities are used to determine Markov chain initial value, we claim that

S is the initial state for Markov chain if $\geq \sum_{i=1}^{S} \pi_i$, S is a random value as $1 \geq \mathrm{S} \geq 0$.

The initial state value represents the current state for FSMC at the first transmission, the future state (previous, current or next) is determined according to the transition probability, which is found from (5.6), (5.7), and (5.8) for the next, previous or current state respectively.

$$P_{k,k+1} \approx \frac{N(\Gamma_{k+1})T_P}{\Pi_k} \quad k=2,3,\ldots, K, \tag{5.6}$$

$$P_{k,k-1} \approx \frac{N(\Gamma_k)T_P}{\Pi_k} \quad k=2,3,\ldots, K, \tag{5.7}$$

$$P_{k,k} = 1-\left(P_{k,k-1} + P_{k,k+1}\right) \quad k=2,3,\ldots,k\text{-}1, \tag{5.8}$$

where $P_{k,k+1}$ is the transition probability from state k to state k+1, which is called next state probability, $P_{k,k-1}$ is the transition probability from state k to state k-1, which called the previous state, and $P_{k,k}$ is the probability to stay in the same state k for the next received SNR value. The future state is chosen to be the state with the biggest transition probability.

$N(\Gamma_k)$ is the level crossing rate for stage k, which is the average number of times per unit interval that the fading signal crosses the signal level Γ_k, and it is given in (5.9)

$$N(\Gamma_k) = \sqrt{\frac{2\pi\Gamma_k}{\gamma_0}} f_m \exp\left(-\frac{\Gamma_k}{\gamma_0}\right). \tag{5.9}$$

From 5.6 to 5.9, the transition probability from stage k to stage k+1, can be approximated by the ratio of the level crossing rate at threshold Γ_{k+1} and the average number of packets per second staying in state k [110].

Figure 5.2 in Section 2.5.4 shows the reason why (5.6) has no next state transition probability for k=K, and (5.7) has not previous transition probability for k=1, as a result, the previous transition probability is set to zero in (5.6) for k=1 and in (5.7) the next transition probability is set to zero for k=K, this makes the current transition probability $P_{1,1} = 1 - P_{1,2}$, and $P_{K,K} = 1 - P_{K,K-1}$.

The final PER is the PER for the current Markov chain state-space, which is calculated from (5.5):

$$PER_k = 1 - \left(1 - P_{ek}\right)^L, \tag{5.10}$$

where L is the number of bits per packet, and P_{ek} is the symbol error rate for state-space k, which is calculated form (5.11) and it is equal to BER for BPSK.

$$P_{ek} = \frac{f_k - f_{k+1}}{\pi_k}, \tag{5.11}$$

where f_k and f_{k+1} are the BER for states k and k+1 respectively, which reflects the channel quality at the state-space time duration.

In our systems BPSK and QPSK modulations, f_k is calculated by (5.12) and (5.13) respectively:

$$f_{k(BPSK)} = 0.5\exp\left(-\frac{\Gamma_k}{\gamma_0}\right).erfc\sqrt{\Gamma_k} + \sqrt{\frac{\gamma_0}{1+\gamma_0}}.\left(1 - 0.5\,erfc\sqrt{\frac{1+\gamma_0}{\gamma_0}\Gamma_k}\right), \tag{5.12}$$

$$f_{k(QPSK)} = \exp\left(-\frac{\Gamma_k}{\gamma_0}\right).erfc\sqrt{\Gamma_k} - \sqrt{\frac{\gamma_0}{1+\gamma_0}}\ .erfc\left(\sqrt{\frac{1+\gamma_0}{\gamma_0}\Gamma_k}\right). \tag{5.13}$$

In conclusion, FSMC realistically models the Rayleigh Fading Channel behaviour as the BER quality depends on Markov chain state-space, which changes with the transition probabilities [110]. Moreover, using our research over Markov chain together with the dividing function makes the change in the transmission distance results to change even in the number of Markov state-space.

Figure 5.2 shows changing of BER f_k according to Markov current state at receiving time from 5e^{-5} till 6.5e^{-4}.

The BER changes according to the time of transmission, though the same transmission power and transmission distance, which makes FSMC practical distance metric.

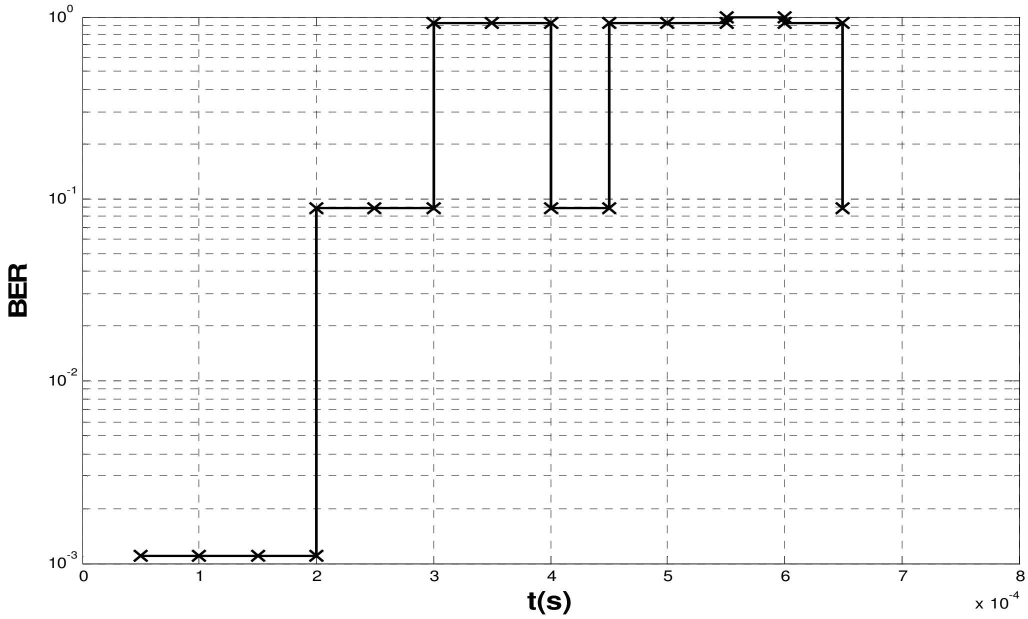


Figure 5.2: shows the changing of BER according to Markov current state-space at receiving time from $5e^{-5}$ till $6.5e^{-4}$.

5.3 Uplink and Downlink Rayleigh Block-Fading Channel

To show the real behavour for the proposed protocols under practical channels, we extend out research over the Rayleigh block-fading channel.

We resort to FSMC which is described in Section (5.1) to be able to convert the transmitted bit streams BER in the physical layer into PER in the MAC layer [110], thus, the channel is modeled with erasures or packet losses.

In the uplink layer section, we apply our proposed NC protocols over WSN system to determine the suitability of using deferent protocols at different conditions, i.e., which protocol is more suitable at different transition conditions.

In the downlink transition, we propose the results for video streams transmission over FSMC in LTE-A to obtain the suitable parameters for different values of PER that can be correctable by the HARQ. So, we assume different level of errors which HARQ can correct, and then find out the QoS for different transmission power for different Relays at different transmission distances.

5.3.1 Uplink System Model

We consider a WSN (or its cluster) that consists of $M \geq 2$ nodes $N_1, N_2, \ldots, N_M$ that transmit equal-length information packets to a common destination node D. The nodes can hear one another as the transmission of different nodes is scheduled over non-overlapping time intervals. For example, users may be co-coordinately scheduled in TDMA fashion by the cluster-head D. We assume that the wireless channel between any two nodes can be modelled as a Rayleigh block-fading channel, where the instantaneous SNR is assumed to be constant during a single packet transmission period. Between packet transmissions, we assume that the channel changes its state according to a finite-state packet level Markov chain (FSMC) model described in [110]. An example of the system model is illustrated in Figure 5.3, where $M = 8$ sensor nodes are placed circularly around the cluster-head D. We use the circular placement model in the following without loss of generality; any configuration of sensor nodes can be analyzed in a similar way as the (Rayleigh fading) channel behaviour (i.e. the average received SNR) depends only on the distance between communicating nodes.

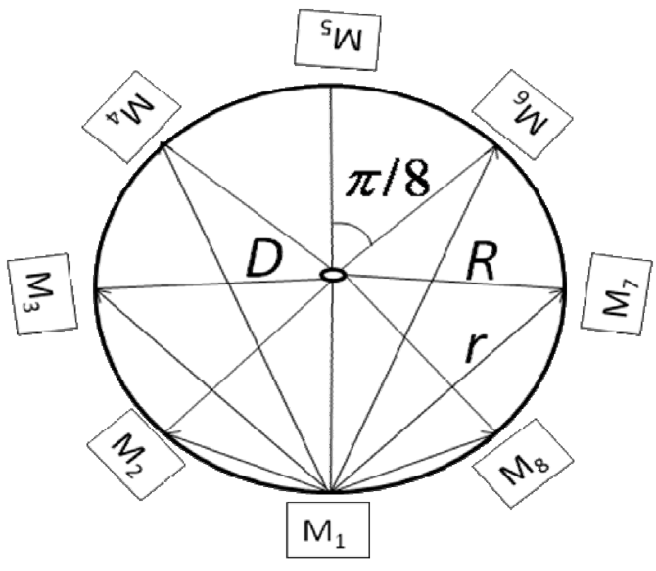


Figure 5.3: WSN example with $M = 8$ Nodes Ni and Destination D, M_1 is chosen to show all distances.

We have chosen this design to obtain different distances between users and then apply FSMC according to the unequal distance. Extensions to more complex transmission schemes that include error-correction coding and/or different modulation alphabets can be easily included as they will only affect packet loss probabilities on Rayleigh block-fading links. The destination informs the nodes with a simple broadcast feedback message when it successfully decodes all M messages. We assume that this feedback message is always reliably transmitted. In the first stage, each node N_i broadcasts its own packet in the corresponding time slot. Each of the remaining nodes in the cluster, that includes the remaining $M-1$ user nodes and the destination node D, receives the packet. The probability of correct packet reception at any of the receiving nodes depends on the instantaneous channel conditions (i.e. the state of the FSMC model) of the corresponding wireless channel, which, on a large scale, depends on the distance between communicating nodes. Stage 1 ends after M time slots.

In the second transmission stage, we propose different random or deterministic cooperative combination strategies with their relative merits and disadvantages as discussed in the following subsections. Our proposed combination strategies can be seen as a simple NC operation over binary field. Indeed, as in NC, a node computes a linear combination of incoming packets. However, in traditional NC [109], each packet is multiplied by a random coefficient from a finite field, and all multiplicative coefficients are sent in the header. In our setup, all multiplicative coefficients are 1, resulting in reduced (for random strategies) or eliminated (for deterministic strategies) need for additional header information. Only one encoded (combined) packet of the same length as the information packets is sent from each node to D. Thus, stage 2 requires M transmission time slots.

After two stages, a total of $2M$ encoded packets, which can be represented as binary linear equations over information packets, are generated by the nodes. When nodes operate in cooperative mode, a maximum of $2M$ different equations can be received at D, but due to transmission over a lossy network, we assume that $m \leq 2M$ different equations are received by D. Since operations are done in the binary field, equations either reveal one unknown (selfish equations) or contain binary sums of up to M

unknowns (combined equations). If, out of m received equations, the number of linearly independent equations is equal M, the destination D can recover all information packets using the Gaussian elimination decoder.

5.3.2 Baseline Non-Cooperative Strategy

For the circular symmetric scenario where users are on the same distance from D, p does not change for different users and can be calculated by averaging the probabilities of packet loss over the states of the corresponding FSMC channel model, and then the same analysis used in 4.3.1 can be applied over this case

More precisely, for the packet level FSMC model containing K states described by the set of steady state probabilities $\pi = \{\pi_1, \pi_2, ..., \pi_K\}$, and the set of packet error probabilities $P_e = \{p_1, p_2, ..., p_\pi\}$ for uncoded BPSK transmission [110], the average packet loss probability $p = \pi P_e^T$. We will use this performance prediction as a baseline for comparison with the cooperative schemes we describe in the next section.

5.3.3 Cooperative WSN Protocol Based on Network Coding

In this section, we heuristically introduce several network coded cooperation strategies in the same way we proposed protocols over the erasure channel, and in the next section, we evaluate their performance in a realistic simulation setting. Before proceeding, we introduce several notions we will use for description of our cooperative protocols. The next neighbour of the node N_i is the node N_{i+1}, with an exception of the node N_M whose next neighbour is N_1. Similarly, we define the previous neighbour of a node. We assume that node indexing is done during the cluster formation and is coordinated by the cluster head node D. The nearest received neighbour of the node N_i is the node or set of which are closest in terms of distance from N_i and from which the node N_i has correctly received a packet in the previous transmission stage. We assume that a node can use Received Signal Strength Indicator (RSSI) to determine the set of nearest received neighbours.

In a similar way, we define the farthest received neighbour or the set of farthest received neighbours (if there is more than one). The packet header is additional and usually very small information content appended to encoded packet to signal certain additional information about cooperative behaviour from each user to the destination node *D*. For each cooperation strategy, we assume that the header content is defined in advance as part of the protocol definition, and is known by both the user nodes and the destination node *D*.

5.3.3.1 Next/Previous Neighbour Combining

The proposed cooperative strategies differ only by the node behaviour during the second stage transmission; the first stage assumes selfish transmission by all the user nodes. For this strategy, if the node fails to receive any of the next or previous node's packets after the first stage, it retransmits its own packet (i.e. remains in the selfish mode). If the node receives either the next or the previous node's packet, it combines the received packet with its own and broadcasts the encoded packet during the second stage. If the node receives both the next and the previous user's packet, it combines all three packets for the second stage as shown in (5.14).

$$C_{r(i)} = \sum_{j=i-1}^{i+1} X_j, \tag{5.14}$$

where $C_{r(i)}$ is the combined transmitted packet at N_i, and $X_j = 0$ if the user N_i does not decode the packet sent from the user N_j. Two-bit binary header field is sufficient for the node to signal the destination which of the four possible second stage behaviour it has applied.

5.3.3.2 Nearest/Farthest Neighbour Combining

In this strategy, the node inspects the set of received packets after the first stage. If none of the packets is overheard, the node remains in the selfish mode during the second stage. If the set of received packets is non-empty, the node combines its own packet with the packet received from the nearest (farthest) user. If more than one of nearest (farthest) user's packet is received, one of the nearest (farthest) users is randomly selected. The motivation for the farthest user selection is that it should increase the diversity of created equations, as the farther the nodes are, the less is the probability they will overhear each other's packets. The second phase transmission requires $\log2_M$-bits header field to explicitly identify the node whose packet is included in the second stage combination; if the node remains selfish, it can simply signal its own $\log2_M$-bit identifier (each of M nodes can be uniquely identified by $\log2_M$-bit identifier).

5.3.3.3 All Received Packets Combining

Same strategy used in (4.3), as the node combines all the packets received after the first stage and broadcast the combined packet during the second stage. Unlike the nearest/farthest strategy, where up to two information packets are combined in the second stage by each node, in this strategy, the node may combine any number of up to M information packets during the second stage.

To describe the encoded combination transmitted in the encoded packet, the node appends M-bit header field in which the i-th bit is set to one if the information packet created by the node N_i is included in the encoded packet.

5.4 Simulation Set-Up and Results for Rayleigh Fading Channel

Consider N users arranged around a circle with radius R from the destination and where the angular separation between users is π/N radians as shown in Figure 5.4, which shows the block diagram for the system design as the FSMC represents the Rayleigh fading channel and the state-space reflects the BER quality.

Figure 5.4 shows the whole system algorithm starting from the transmission power for the first stage.

The algorithm works from the transmission values, which is assumed to be the same for all users at all stages. This P_T power is transmitted to the N users and D, to be received as $P_R(R)$. The path losses is calculation reflects the distance Matrix, i.e. the received SNR Matrix is related just to the distance as the transmitted power is assumed to be the same for all users. Moreover, Figure 5.4 shows where the retrieving step is located in this scenario which is the main feature gained by applying NC cooperation.

To differ the transmission between users themselves and users with D, we arranged the distance matrix to have R diagonal and we assumed the diagonal SNR received values are the values sent from the N users to D, for example, the first element of the SNR $R_{r(1,1)}$ received Matrix represents the received SNR value from user one to D, and $R_{r(2,2)}$ represent the second user to D and so on.

The received SNR values are fed into the dividing function to calculate the number of space-states and the partition values for each received value, taking into consideration that C_k is assumed to be fixed for the all received SNR values.

As described in section 2.5.4, FSMC needs just the number of the state-space and the received SNR partition values to represents the channel quality BER at the time, and to determine the future state, taking into consideration that the probability and future states Matrix's diagonal represents the N users to D transmission.

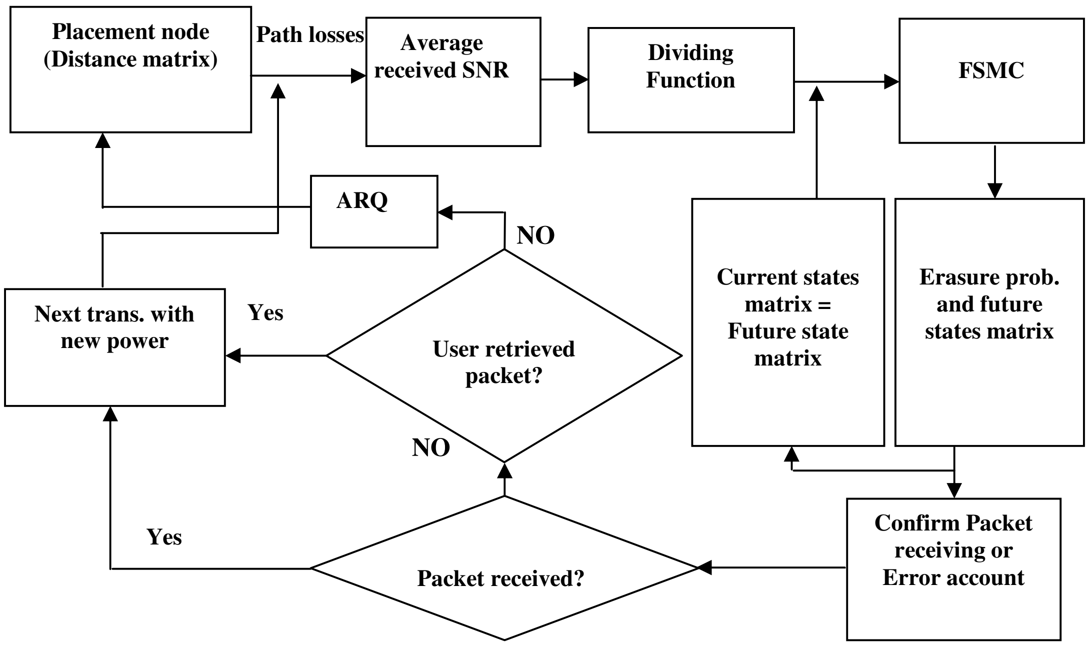


Figure 5.4 System set-up Block diagram.

At the end of the stage, we evaluate the received packets, and in the case of all packets received, the algorithm ends by sending the future states to the FSMC for the new transmission, and an acknowledgement message sent from *D* to the N users confirming the end of the algorithm, otherwise, the next stage follows, where users apply NC over the overheard packets between them and re-transmit the combined packets.

As the added part of the combined packets is neglect able, we assume that the same power transmission for the combined packet.

At the same time, the future states Matrix is fed to FSMC to be the next current state which used to determine the new future state-spaces for the transmission after the next.

5.4.1 Simulation Results

In this section, we present simulation results for the proposed node cooperation schemes based on NC introduced in the previous section. We focus on a realistic modeling of a circular cluster WSN topology with M = 8 user nodes and the destination node *D*, as presented in Figure 5.3. The radius of the circle, i.e., the distance between the user nodes and the destination node *D* is r = 25 *m*. The users are regularly placed on a circle with an angular distance between the neighbouring nodes equal $\alpha = 2\pi / M = \pi / 4$ rad.

The carrier frequency f_c = 2.4 GHz. The nodes transmit information/encoded packets using uncoded BPSK signaling where the transmission rate is set to R = 100 kbit/s and the bandwidth of the transmitted signal is BW = 50 kHz. The equal length information/encoded packets are of size L = 500 bits, which makes the duration of a single packet transmission equal $T_p = L / R = 5$ *ms*. The power of MICAz nodes is changed from P_T = −15 *dBi* to P_T = 5 *dBi* with the step of 5 *dBi*. The gain of both transmit and receiving antennas is assumed to be G_T = G_R = 5 *dBi*. The path loss between sensor motes follows the well-known Friis path-loss law [111]:

$$P_R(R) = G_T G_R .(\lambda / 4\pi)^2 (1/R)^n .P_T, \tag{5.15}$$

where $\lambda = c / f$ is the carrier wavelength and the path-loss exponent which is set to n = 4.4 [112].

In the WSN cluster configuration, all nodes are considered to be static, which is why, once the motes are fixed, the channel gains could be considered time-invariant. However, as we are mainly interested in outdoor WSN applications, to realistically model the channel behavious, we assume very slow fading process on each of the channels between motes as a consequence of the motion of surrounding objects. We model these changes by a packet-level FSMC channel model based on the equal average duration channel state modeling [110].

The average duration the channel spends in any FSMC state is set to C_k = 15 packets and the product that characterizes the fading speed of the channel relative to the packet length is set to $f_D T_P = 0.01$ [110].

As the distance, transmission power and the number of users are the most important parameters in WSN design, we have investigated the protocol's behaviour of increasing one parameter while the other two parameters are remain fixed.

Figure 5.5 shows how increasing the transmission power in two stages transmission improves PER significantly, moreover, it shows how our proposed strategies outperform the selfish strategy when NC is not applied, in fact, and it does outperform even the random combined strategy.

So, at *PowTx* =-9 *dBi*, PER in the selfish is as low as 0.5, compared to 0.0033, 0.0037 and 0.18 for all received combine, next neighbour only combine and random combine protocol respectively. In Figure 5.6, we fix *PowTx* at -9 *dBm* and r at 25 meters, which gives best average results, and then the number of users *M* increased from 6 to 14.

We can notice that increasing the number of users has lemited reverse reflect over the results, which is clearly justified by the fact that the path losses depends on the distance, however, there is PER noticibale losses when we increased the number of users from 6 to 14 in all received combine protocol and selfish mode, for example, increas *M* from 6 to 14 results to decreasing PER from 0.0008 to 0.0036 in all received combined protocol.

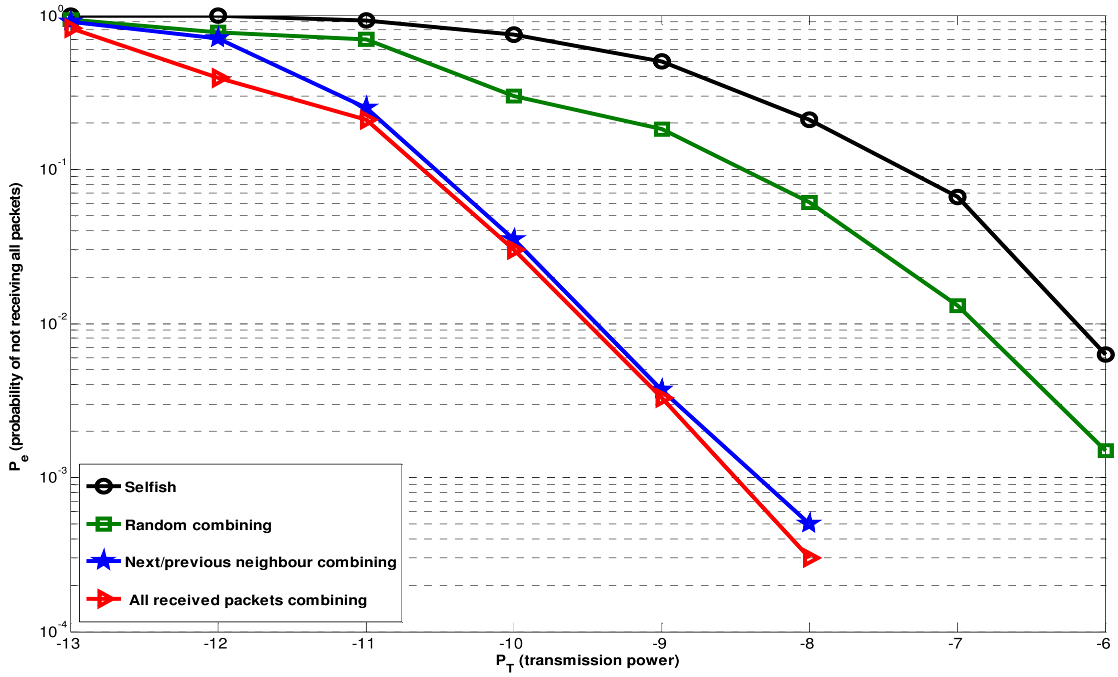


Figure 5.5: Three stages Network coding strategies for different values of *PowTx* at *r*=25 and *M*=8 compared to the selfish strategy.

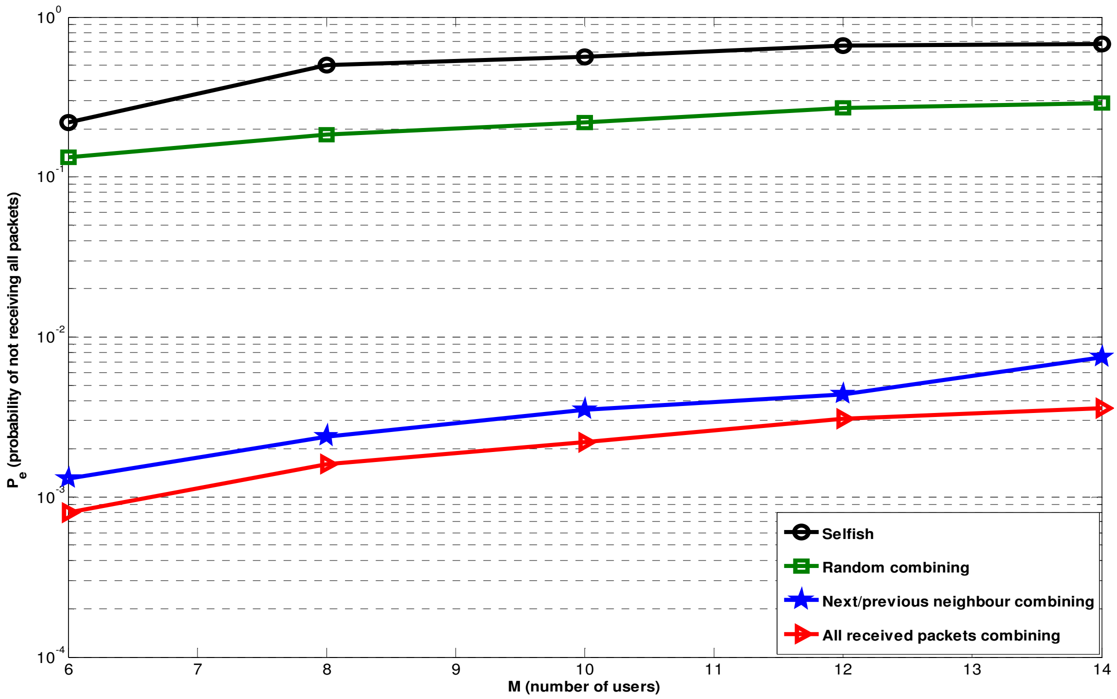


Figure 5.6: Two stages transmission for *M*=6, 8, 10, 12 and 14 for the proposed strategies at *r*=25 and *PowTx*=-9 *dBm*

The most important factor in WSN is the distance when fixing the transmission power, as the path losses law has a reverse relation to the power value of the distance as in (5.15). Whish is the reason why PER decreases from 1e^{-4} to 0.59 in all received combined strategy when just increasing the transmission distance from 23 to 30 Meters at M=8 and $PowTx$=-9 dBm

Moreover, the severe PER decreasing versus the distance is justified by not applying any channel coding in our results, as shown in Figure 5.7.

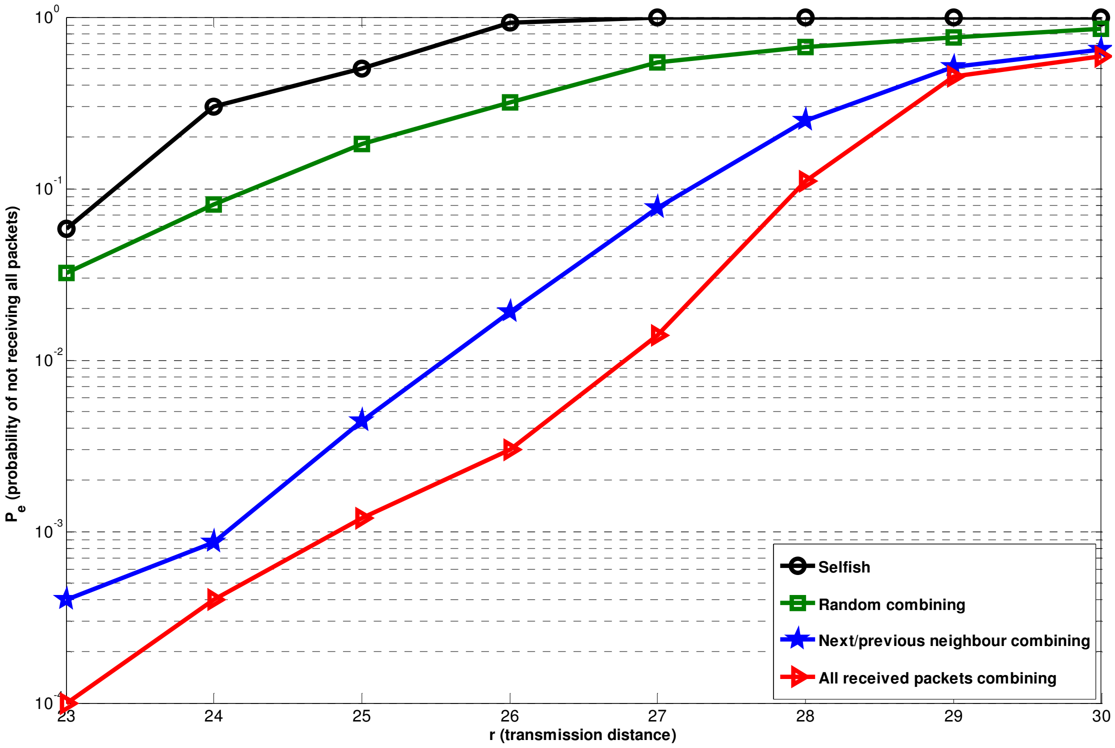


Figure 5.7: Two stages NC strategies for different values of r at M=8 and $PowTx$=-9 dBm.

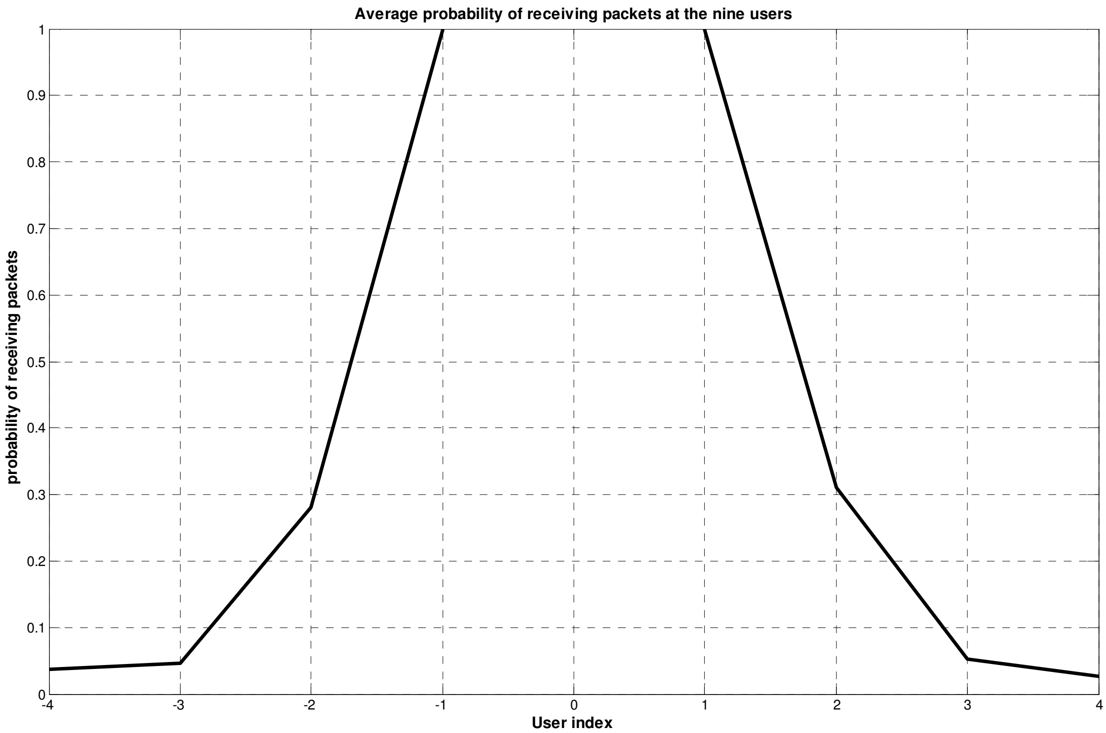


Figure 5.8: Probability of receiving packets from the node i ± n, n = 1, 2, 3 and 4, for 10000 transmission times.

Figure 5.8 shows that the next neighbour is the most combined packet in combined all packets protocol, and then the nearest. User 5 for example always receives packets from its immediate neighbours, users 4 and 6, and combines them before forwarding to D. However, the rate of reception and combination of packets drops to about 26% from neighbours 3 and 7, 4% from neighbours 2 and 8, and 2.17% from its furthest neighbour (user 9). These results show why the 'next user' protocol is as good as the 'combine all' protocol, as most of the combined packets in the combine all protocol are the next users

From the example of user 5, we conclude that each user gives the combination priority to its immediate neighbour. All users exhibit similar performance since the model generating the outage probability is only a function of distance (when P_T is fixed) and all users are currently arranged in a circular fashion. However, we can generalize the distance to be the key factor in determining outage probability and hence performance for any topology of users. We also observe variability in rate of packet reception (and hence combination) for the immediate neighbour (next and previous), though they are located at the same distance from the receiving user, such as in user 5, where the reception rate for user 7 is 26% but that of user 3 is 31% . This is the result of the variability of the model justified by the changes in the FSMC state-space behaviour, which reflects the BER quality for the channel at the transmission time.

5.5 LTE-A Network over FSMC for Downlink Scenario

In this section we introduce the downlink behaviour for video streams transmission over block-fading channel, which is represented by FSMC as in the downlink scenario which is shown in section 5.3.1.

The scenario under investigation in this section is designed for LTE-A system which implements one Pico and two Femto relays to transmit video streams of data to a mobile user, which is the same scenario proposed for the downlink scenario under erasure channel in section 4.4.1 and shown in Figure 4.5.

We restrict our research over the influence which HARQ can provide to the system by the ability of correction different levels of PER. Moreover, we show the relation between the transmission distance, transmission power and QoS accordingly.

In addition, in this section we modify FSMC presented in [110] to be able to deal with mobile nodes over two different speeds (3KM and 30KM) and we show the received data rate for the system after certain number of transmission rather than showing only the transmission data rate in the case of the tradition FSMC, to be another obtained improvement.

5.5.1 Introduction

Recently, a lot of attention has been paid to video transmission over LTE-A network, benefiting from Release 10 that provides HARQ and CoMP features that decreases the ARQ and supports cooperation over the network.

Based on above, number of researches that apply NC over transmitted video streams exploiting CoMP has been raised significantly to improve the network diversity. This improvement in the diversity has enhanced the quality of service in video communication systems, resulting to better bandwidth throughput and packets losses.

To achieve such desirable improvement efficiently, appropriate streaming mechanisms plays a vital role to fully exploit the network resources and to minimise the redundancy [113].

In [6] NC has been applied over the video streams via an intermediated relay. In such case, the relay performs NC by combining the incoming messages rather than forwarding them with no process to the destination. Applying NC over multimedia applications however suffers from lack of simplicity and time delay constraints imposed by practical applications. So, it is important to adapt NC technique for multimedia streaming to obtain realistic networks that cope with streaming specific requirements such as delay constraints.

Reed-Solomon codes or digital fountain codes were among the earliest researches that have been implemented in network-embedded FEC nodes [114] and in network peers [115] showing that NC improves the throughput significantly mainly when the packets are decoded and re-encoded before forwarding to the destination, though latency occurred to the streaming system.

In [116] re-encoding packets with rateless codes have been proposed especially for high loss rate erasure channels with limited diversity to be robust to erroneous channel estimations.

Though [117] introduces a design of an efficient device-to-device communication mode which is not the scenario we are investigating, but it does perform the communication underlay to a cellular network over LTE-A Release 10. This work showed us the resent parameters for Release 10 being used, which is good to understand how LTE-A behaves under such parameters.

To compare different NC scenarios, [118] and references there in; give a wide tutorial work with specific focus on multimedia streaming, based on NC to allow nodes to create and forward "combinations" of incoming messages to increase throughput, such as [119] which proposes NC for Line media streaming over Peer-to-Peer communication, and [120] which introduces opportunistic NC for Video Streaming over Wireless.

The reason we are proposing FSMC in this work at first place is to understand the LTE-A and then have the basic material to evaluate applying NC over LTE-A in future work.

FSMC channel has been simulated to represent a functional distance merit and more realistic erasure probability for the scenario under investigation for wireless networks over MDC and RLC LTE-A network.

We use FSMC to model transmission video over LTE-A channel with the latest in Release 10 aspects such as 2.2GHz carrier frequency and 1000 Kbps data rate.

5.5.2 Finite State Markov Chain Received Data Modification

In our collected results, we implemented FSMC over the LTE-A as a practical distance factor to represent the block-fading channel. Moreover, we have modified the channel to adapt with the mobility of the users resulting to different received data rate.

Based on above, FSMC practically models the Rayleigh Fading Channel behaviour as the BER quality depends on Markov chain state-space, which changes with the transition. As it is fact that some packets are being lost in the erasure channel, the received data rate will be less than the transmitted; in our work we did take this fact into account when designing our FSMC channel by applying equation (5.16) over space-states:

$$R_{DR(k)} = \begin{bmatrix} PER_k \geq HARQ\_CL & \dfrac{L}{T_P} \\ PER_k < HARQ\_CL & Zero \end{bmatrix} \tag{5.16}$$

Where $R_{DR(k)}$ is the received data rate for state k, and HARQ_CL is the HARQ Correction Level, which is the level of PER that HARQ can correct without the need for the ARQ. i.e., it is the error probability that each received packet is assumed not to pass. If any packet has error probability more than HARQ_CL, this packet will be assumed as not received. So, (5.16) shows that if the PER for state k (PER_k) is less than the HARQ_CL, i.e., the packet will be received, then the received data rate will be added to the total received data, otherwise, the received data rate for this state will be zero as this packet will be assumed not received. As a result, the total received data rate depends in the HARQ_CL and the time of transmission (space-state PER).

Total received data rate after a certain number of transmissions (Trans-No) assumed to be is shown in (5.17):

$$R_{Total-DR} = \frac{\sum_{i=1}^{Trans-No} R_{DR(i)}}{Trans-No} \text{ bps.} \tag{5.17}$$

Where $R_{DR(i)}$ the received data rate for transmission time slot *i* which is shown in (5.16) and it depends on the FSMC reception state and HARQ_CL PER.

Our results have been collected under slow and medium speed. In the case of the medium speed, we confine our results to represent the transmission distance with the transmission power when full reception is obtained.

5.5.3 Results and Analysis

In this part we show the relation between the transmission powers; as it is the major factor in FSMC and the received packets for the scenario adapted in this work for all paths. The paths which we have to test are from the Pico relay to the two Femto relays and the user. Moreover, we show the change in received data rate with transmission time, i.e., time varying received data rate according to FSMC when changing the transmission power as our modification to the standard FSMC [110] as shown in (5.17). In addition, unlike the standard FSMC which is proposed for static users, we used the f_D for movable users at low and medium speed taking into consideration (5.19) which is same in fading channel (2.8):

$$f_D = \frac{v}{Cf_c} \ Hz \tag{5.18}$$

Where v is the vehicle speed (m/s), C is the light speed and f_c is the carrier frequency.

The two speeds we used are low speed at 3KM and medium speed at 30KM, which means the two f_D used are 5.5 Hz and 55.56 Hz.

The simulated packets transmission over FSMC in LTE-A has been simulated as follows:

- Set up the system parameters such as transmit power, path loss, transmission distances, bandwidth, data rate and Doppler shift.
- Determine the number of transmitted packets to run the simulation over.
- Dividing Function shown in Figure 5.4 is used to determine the optimum SNR intervals and state\transition probabilities for FSMC.

- The FSMC then starts off in a random starting state, which is found according to the steady state probability that reflects the randomly chosen number between 1 and 0.
- The FSMC keeps changing the states throughout the transmission to represent the time varying BER quality for the transmitted packets as in [110] which is used in Chapter three simulations.
- PER is obtained from the stage the packet is located at, ending up with full PER record for all transmitted packets.

The parameters used in the FSMC are the ones used for LTE-A that specified by Third Generation Partnership Project (3GPP), which are 46 dBm transmit power for the base station, 14 dB antenna gain for the base station, 36 dB. Transmission and receiving antenna gain is 5, -100 dBm/Hz Noise power, 20 MHz Bandwidth, and 2GH carrier frequency [52]. Then, we chose the packet size to be 6352 bit and the data rate to be 1Gbps. As a result, the packet time will be the packet size over the data rate which is equal to $6.3e^{-3}$ s. To assure stable results, we run the code for 500000 transmission times, which means the minimum transmitted packets are 500000.

We need to modify our FSMC used in the static case to be capable for LTE-A, mainly in three issues which are:

- LTE-A model works with changeable number of users (nodes).
- LTE-A users i.e., UE, are dynamics, so, they are moving with high speed.
- The path loss is calculated in different way than Friis path-loss law (5.15).

Path loss in LTE-A refers to the signal attenuation as a function of distance, so, the received SNR is calculated according to the link budget equation (5.19), which called budget equation as the calculation is set out in much the same way as a financial budget, taking in to account the various sources of gain, attenuation and noise [121]:

$$SNR_{db} = P_{TX} + G_{TX} - P_L(d) - NPSD \qquad (5.19)$$

Where:

- P_{TX} = transmitter power
- G_{TX} = transmitter antenna gain
- G_{RX} = receiver antenna gain
- $P_L(d)$= distance dependent path loss
- NPSD = Noise Power Spectral Density

The manner the path loss varies with distance depends on the particular model used, such as Okumura-Hata or Walfish-Ikegami [122]. The path loss models used in this contribution conform to those suggested by the 3GPP for system level simulation [122]. Table 5.1 lists those used in this application.

Table 5.1: Path losses assumed in LTE-A

Macro to UE	$128.1 + 37.6 * \log_{10} R_{km}$
Femto to UE	$145.4 + 37.5 * \log_{10} R_{km}$
Macro to Femto	$131.1 + 42.8 * \log_{10} R_{km}$

Where Macro is a conventional base station that uses a dedicated backhaul and opens to public access, with typical transmit power for Macro ~43 dBm; and the antenna gain is ~12-15 dBi. And Femto as a consumer-deployable base stations that utilize consumer's broadband connection as backhaul, with typical transmit power < 23dBm [123].

The parameters used in the results are 2GHz carrier frequency, two speeds low and medium (3 KMph and 30 KMph) respectively. The Data rate is 1000 Kbps, 450 Kbps and 500 Kbps for the Pico, Femto 1 and Femto 2 relays respectively.

The results show the behaviour of the QoS when changing the transmission power at different HARQ_CLs, i.e., the received packets at different ability of HARQ to correct the received packets over the erasure probability when changing the transmission power. The scenario under investigation is shown in

Figure 5.9 with the real distances used in the simulation results.

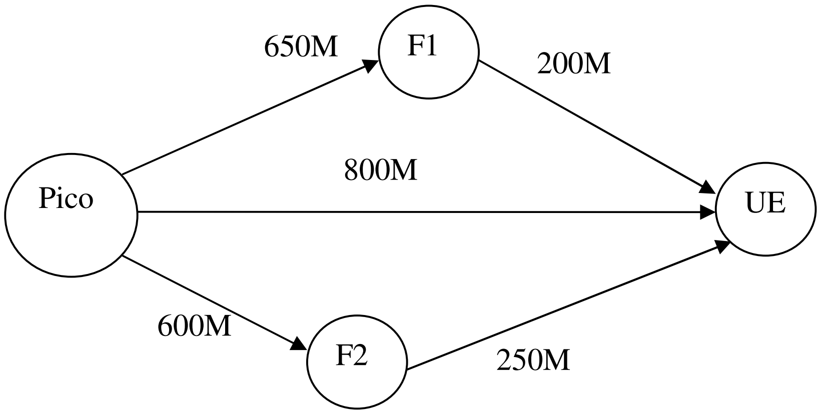


Figure 5.9: The real location for the scenario under investigation.

So, we start with showing the improvement in QoS when HARQ_CL increases from 0.05 to 0.30 as shown in Figure 5.10.

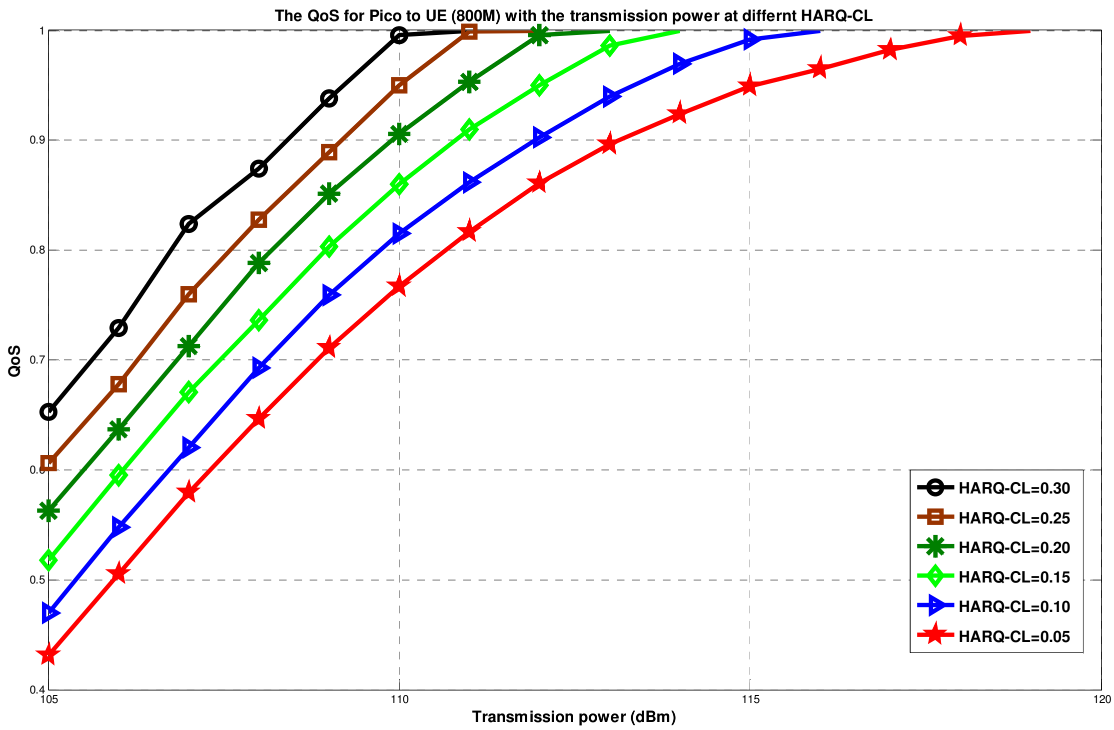


Figure 5.10: Pico to User End QoS at different level of HARQ_CL and versus the transmission power at speed of 30KM.

We can see that the QoS has improved from 0.75 to 0.996 at 110 dBm transmission power when HARQ_CL increases from 0.05 to 0.30 which is justified by the ability of the system to correct the packets rather than considering them as lost, and the same behaviour can be seen at different transmission power.

The other transmission paths have the same influence from improving the error correction level as shown in Figure 5.11and Figure 5.12 for Femto 1 to UE and Femto 2 to UE respectively.

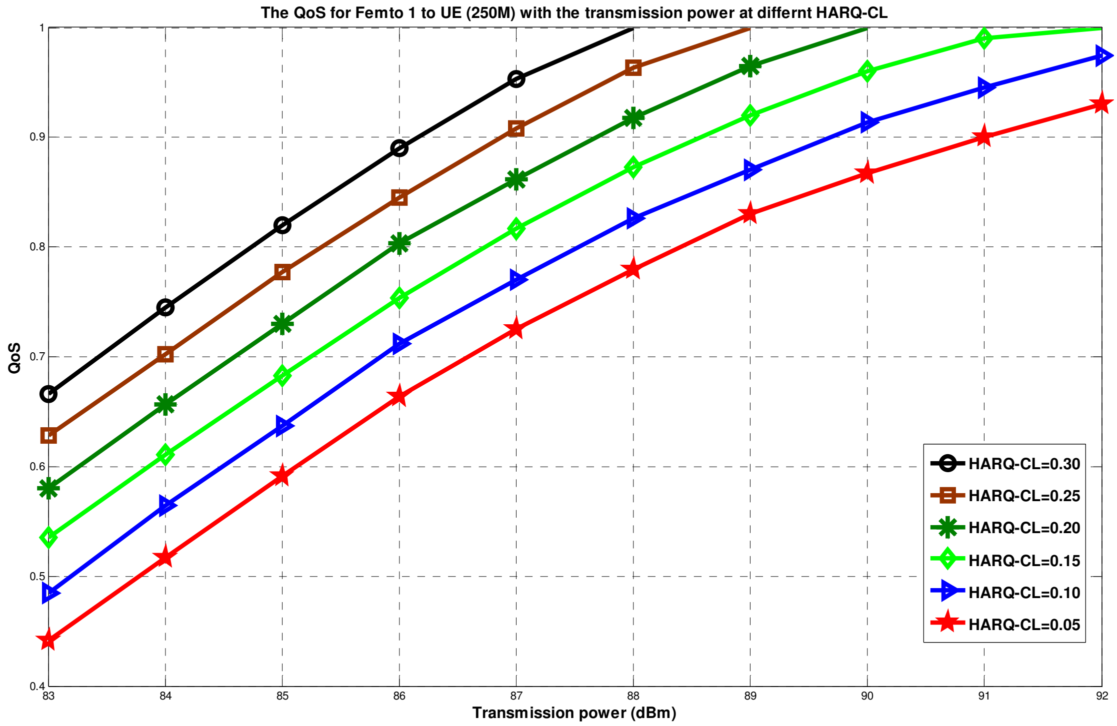


Figure 5.11: Femto 1 to User End QoS at different level of HARQ_CL and versus the transmission power at speed of 30KM.

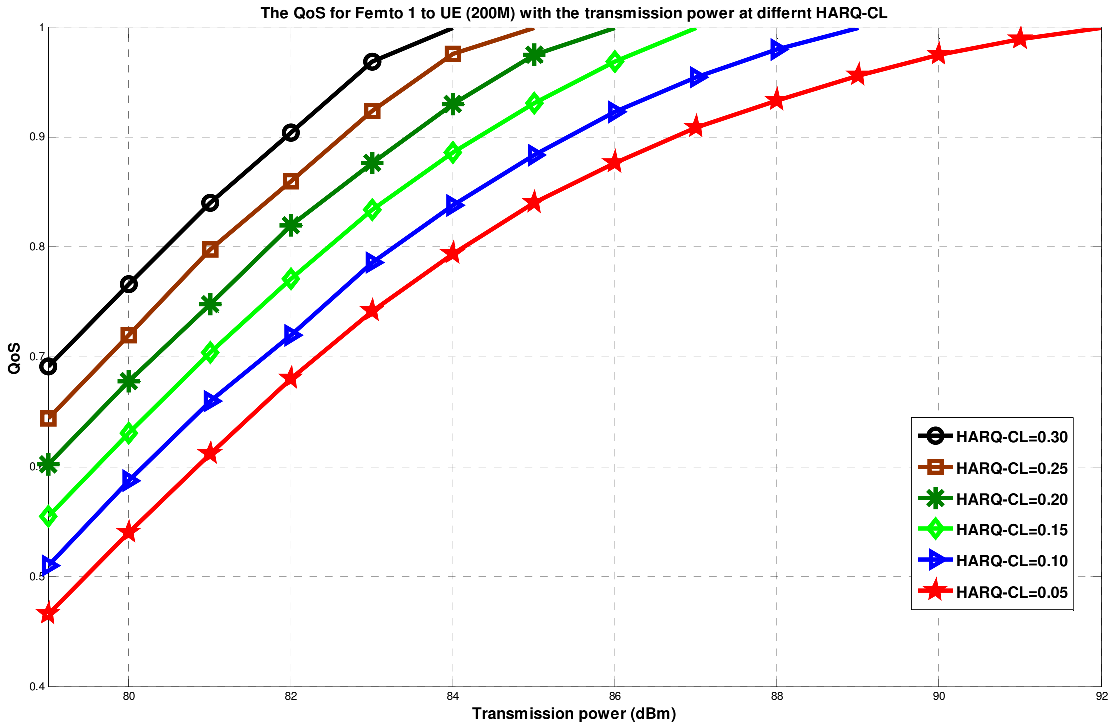


Figure 5.12: Femto 2 to User End QoS at different level of HARQ_CL and versus the transmission power at speed of 30KM.

When we change the distance between the Femto relays and the UE to higher than 200M and 250M, we still have the same behaviour, which shows that the system is table while the UE is moving with 30KMPS.

Figure 5.13 Figure 5.14 shows the improvement in QoS when we increase the two Femto relays to 600M and 650M respectively.

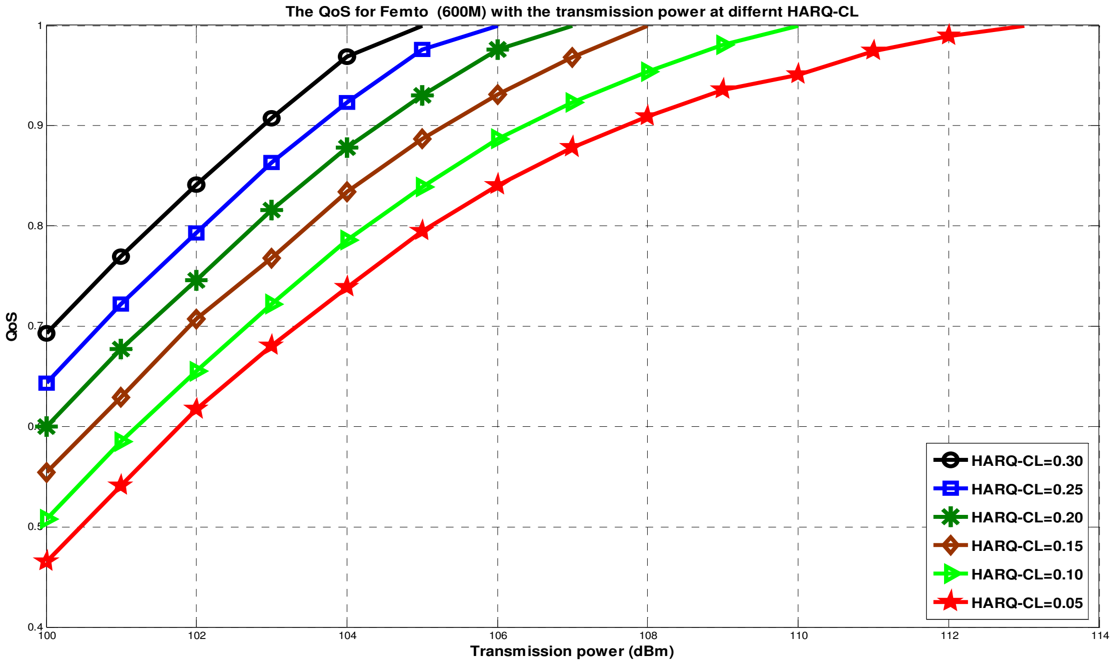


Figure 5.13: Femto 2 to User End QoS at different level of HARQ_CL and versus the transmission power at speed of 30KM.

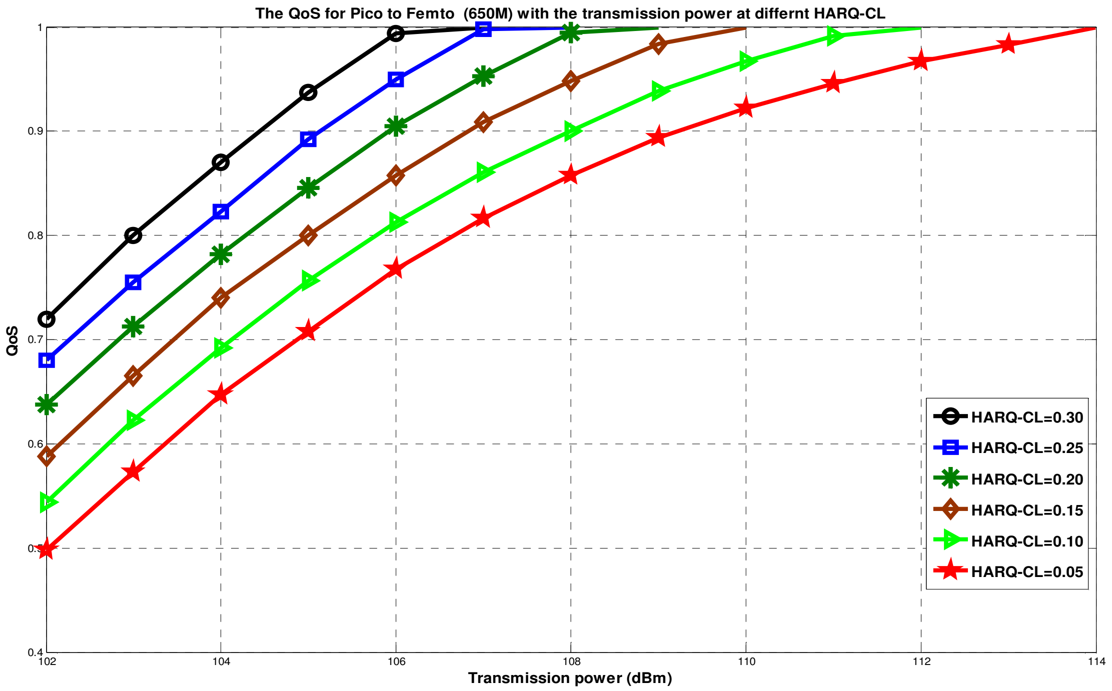


Figure 5.14: Femto 2 to User End QoS at different level of HARQ_CL and versus the transmission power at speed of 30KM.

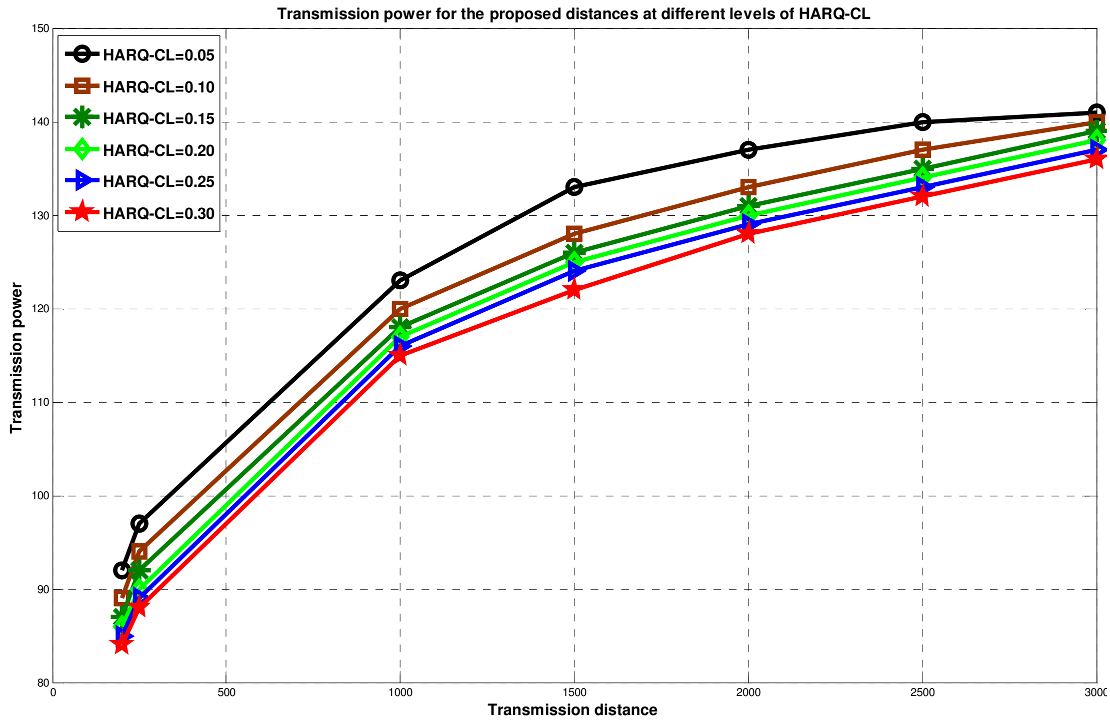


Figure 5.15: Different transmission distances at different level of HARQ_CL at speed of 30KM.

The other part of the results is to show the behaviour when we change the transmission distance at a certain HARQ_CL or vice versa to obtain full reception i.e., QoS=1, as shown in Figure 5.15.

The figure shows that we need 128 dBm transmission power to obtain full reception at HARQ_CL=0.30 compared with 137 dBm transmission power at HARQ_CL=0.05.

This example shows us how HARQ_CL saves the transmission power at a fix transmission distance, or improves the transmission distance at a fixed transmission power.

Finally, to show the trend of the reception packets in FSMC at 30KM, Figure 5.16 shows the received and not received packets in a random results round.

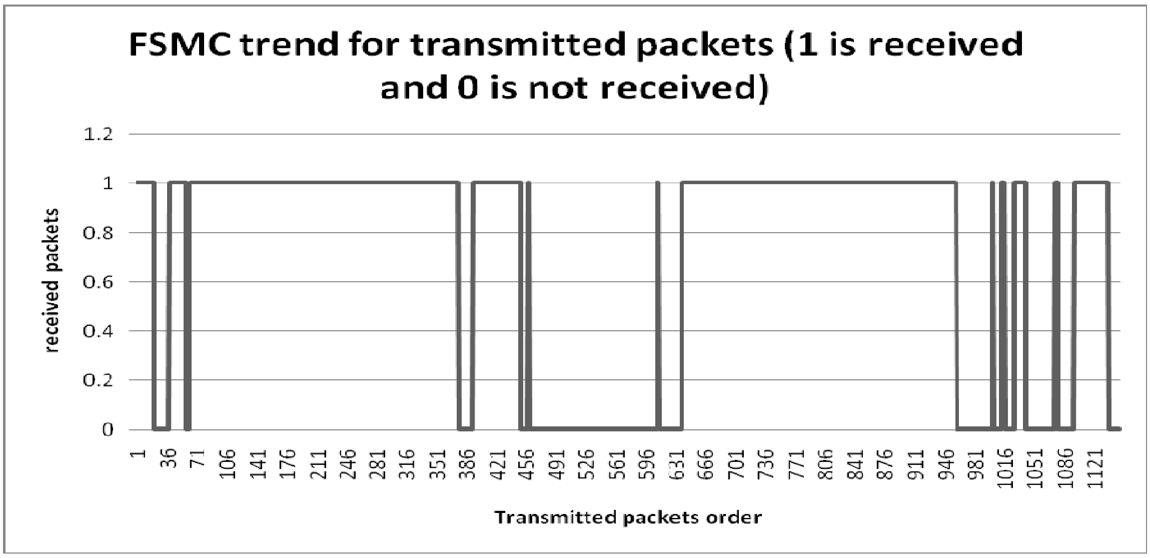
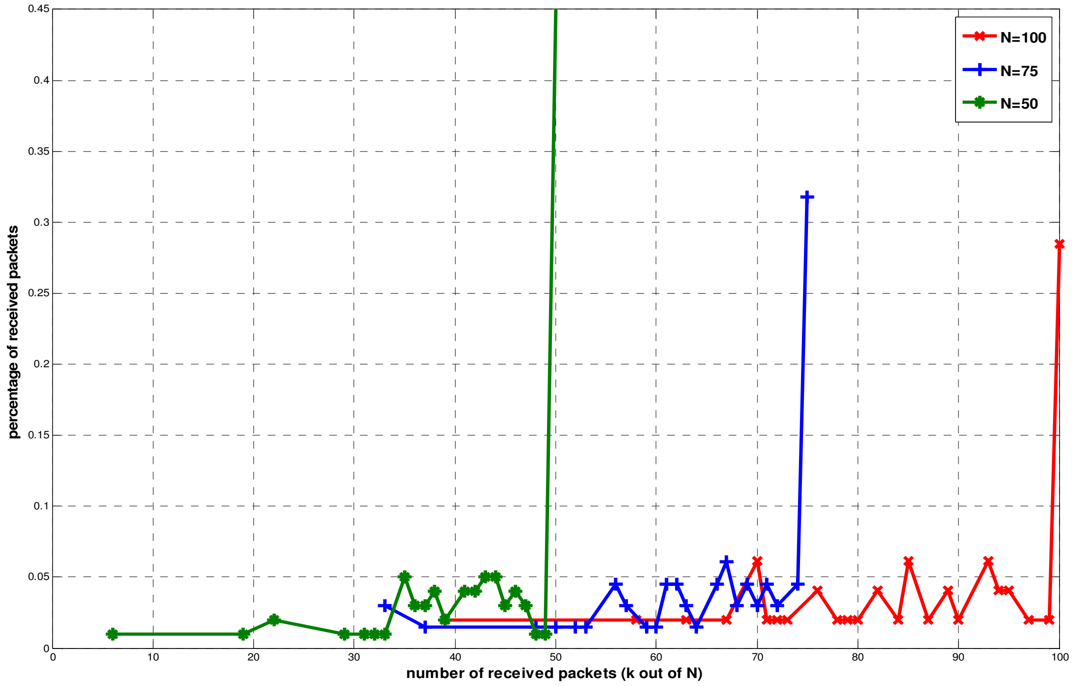


Figure 5.16: Received packets over FSMC channel at the user side, Figure 5.16 reflects the error burst in FSMC channel.

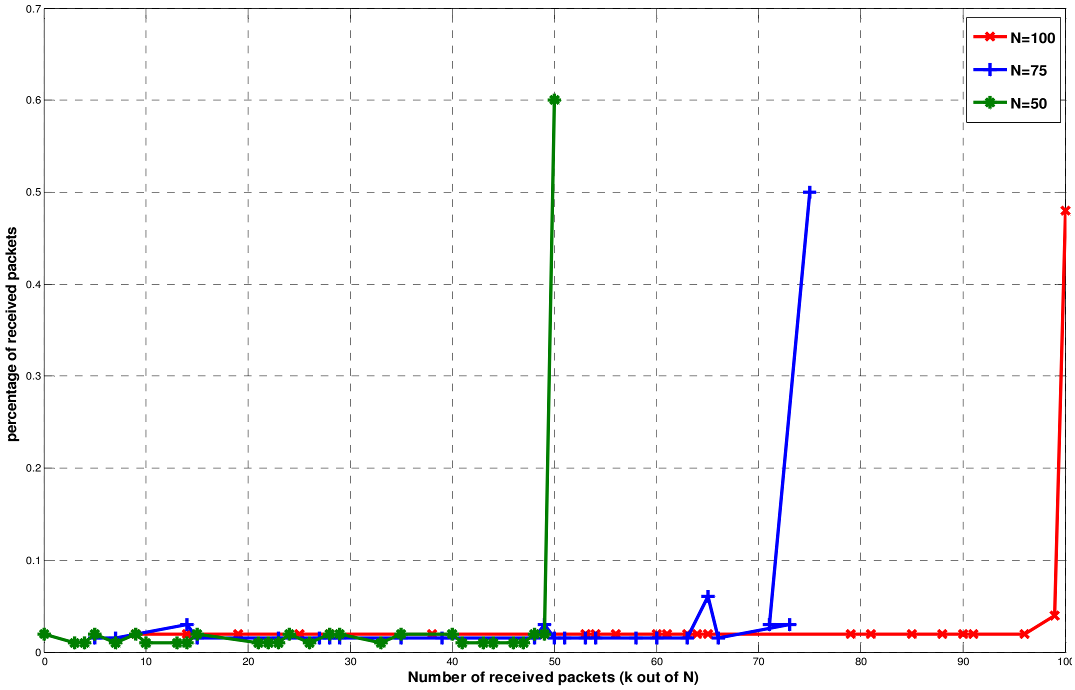
Figure 5.16 shows that in the used number of transmitted packets we have several burst errors through the transmission. The size of the burst error determines the used combination. According to Figure 5.16, we can notice that we have one big burs error in the packets between 456 till 596. This big number of burst error forces us to chose to less combined packets protocol which is two packets, moreover, in (4.20) shows that the combined packet should be as far as possible to jump the burst error, in this case, *b* in (4.20) should be more than the gap's size which is in this example 140.

In the other hand, we have such big 100% received packet areas in the Figure 5.16, particularly from 71 till 386 and from 631 till 946, so, using *k*-1 deterministic protocol is recommended.

Finally, we show the Probability Density Function (PDF) figures to show how our results are matching the exponential channel attenuation for FSMC, showing the PDF for two different QoS at 0.70 and 0.95 as shown in Figure 5.17 (a) and Figure 5.17 (b) where the figure represents three differences of transmission groups which are 100, 75 and 50 packets each. The x-axis shows the percentage of the received packet numbers per total transmitted for the three different groups. The y-axis represents the number of packets received out of the total transmitted packets.



(a)



(b)

Figure 5.17 : (a) PDF for QoS= 0.70, and (b) QoS =0.95.

Figure 5.17 (a) and (b) shows the PDF for FSMC at QoS of 0.70 and 0.95 respectively. We can notice that FSMC follows the exponential behaviour for the received packets as we can notice from Figure 5.17(a) where almost all packets percentage received are over 70% and the reception percentage drops exponentially which is the same for Figure 5.17 (b) where almost all packets percentage received are over 95% and then the reception percentage drops exponentially as well.

5.6 Conclusion

In this chapter we have extended applying Network Coding (NC) over the Finite-State-Markov-Chain (FSMC) erasure channel.

In the first part of this chapter, our research has been extended to use FSMC as a functional distant merit over a cluster of WSN nodes and further protocols have been investigated over the proposed practical transmission protocols for a typical localized data gathering scenario within a WSN cluster mentioned in Section 4.3.

Implementing FSMC to represent block Rayleigh fading channel showed us the most suitable combination for WSN in the practical design.

Moreover, we derive practical analysis for the probability when users are in different destinations from each other and the same distance from the destination. Results show the probability when the destination does not recover all users' packets, after one and two transmission stages adopting our proposed cooperative protocols based on deterministic and combine all received packets NC. Simulation results demonstrate the performance of the system when manipulating the transmission power, transmission distance and number of users (cluster) which shows a perfect match with the analytic results for the path losses equation and also show the performance benefits of using deterministic over random combination strategies.

In the second part of this chapter, we implemented FSMC over Long-Term-Evaluation-Advance (LTE-A) network for the scenario of one Pico and two Femto relays to show the practical results for the scenario under investigation. Moreover, we showed the behaviour of LTE-A when changing the HARQ correction level with the received Quality of Service at different level of transmission power and distances.

Modifying FSMC [110] to be able to show the received data rate and to work under movable nodes instead of just statics' has been introduced as a part of the novelty.

Finally, the implemented FSMC has been investigated itself, by showing its exponential receiving rate for different groups of transmitted packets. Moreover, the changing in reception packets per transmission time was introduced to help choosing the proper NC for further research over LTE-A.

6. Conclusion and Future work

6.1 Thesis Conclusion

In this thesis we focused our contribution on applying cooperative Network Coding (NC) over wireless networks. We started our research by applying a low-complexity physical layer network encoding and decoding scheme for bandwidth- and power-savings for an information exchange scenario via a relay with Amplify-and-Forward (AF) or Decode-and-Forward (DF) based schemes. The systems combine NC with high-performance partial-unit memory-based turbo codes for forward error correction. The theoretical limits of capacity for the proposed schemes are shown with the schemes' behaviour in high and low SNR regimes. We applied a deterministic combination scheme where messages from two nodes are combined by the relay before broadcasting by AF or DF, yielding a savings in $1/N$ in transmissions. A modified version of the Gauss-Jordan elimination algorithm is proposed for message recovery exploiting the system set-up. We propose broadcasting additional packet combinations to decrease the effect of error propagation inherent in the recovery process. Simulation results for all proposed schemes demonstrate their relative performance over the benchmark scheme, and are promising due to their performance and simplicity.

After testing the results over the physical layer, and motivated by the good FEC code used, we extended our research over the MAC layer to improve the diversity which results to improve the communication between the users.

So, we introduced NC in cooperation transmission protocols for two different scenarios, a typical uplink localized data gathering scenario within a WSN cluster and over LTE-A downlink three relays network scenario. Both scenarios boast simplicity and low-complexity while still maintaining performance gains over non-network coding solutions.

We discuss analytical and simulation results for the probability in the uplink scenario that the destination does not recover all nodes' packets, after two and three transmission stages adopting our proposed cooperative protocols based on low-complexity binary field NC. Simulation results demonstrate a perfect match with the analytic results and demonstrate the performance benefits of the proposed protocols over the baseline repetition protocol. In the downlink LTE-A scenario, we adapted the simulation results to illustrate our analysis and to present the improvement gained by applying NC over the Femto relays.

Moreover, we point out that the deterministic combining protocols are as competitive as the non-deterministic, but without the header overhead and with simple Gaussian elimination algorithm. So, we showed that full connectivity in both the uplink and downlink scenarios are maintained even when the network drops a significant number of packets through the erasure channel, and we showed that this is justified by the ability of the sink to retrieve the propped packets than requesting them to be retransmitted.

In the last chapter of the research, we have extended applying NC over the FSMC erasure channel as a functional distant merit over a cluster of WSN nodes and further protocols for a typical localized data gathering scenario.

Implementing FSMC to represent block Rayleigh fading channel showed us the most suitable combination for WSN in the practical design. Moreover, we derive practical analysis for the probability when users are in different destinations from each other and the same distance from the destination. Results show the probability when the destination does not recover all users' packets, after one and two transmission stages adopting our proposed cooperative protocols based on deterministic and combine all received packets NC. Simulation results demonstrate the performance of the system when manipulating the transmission power, transmission distance and number of users (cluster) which show a perfect match with the analytic results for the path losses equation and also show the performance benefits of using deterministic over random combination strategies.

In the last part of this chapter, we implemented FSMC over LTE-A network for the scenario of one Pico and two Femto relays to show the practical results for the scenario under investigation. Moreover, we showed the behaviour of LTE-A when changing the QoS and PEP at different level of transmission power, in addition, we showed the recommended transmitted power for different distances at different QoSs.

Modifying FSMC [110] to be able to show the received data rate and to work under movable nodes instead of just statics' has been introduced as a part of the novelty in this chapter. To confirm the correct attitude of the implemented FSM, the exponential receiving rate for different groups of transmitted packets has been obtained.

6.2 Future Work

Generally, our contributions are applying NC and cooperation over the physical layer and upper layers, i.e., we have performed these separately. Our future research is directed to apply NC and cooperation over a two-layer coding scheme where physical layer channel coding is utilized within each packet for error-correction and then the deterministic NC is applied on top of channel coding for network error-control, to increase the reliability of the transmission over the relay system, In such case, PUMTC is proposed to be the FEC physical layer channel coding, after confirming the error free data, the higher layer deterministic NC and cooperation is applied over the packets that considered as error free data packets. The reason behind this direction is to join the benefits which have been obtained over the different layers. As a result, deterministic NC and cooperation is proposed to be applied, and then the channel coding is used to improve the error performance on the physical layer. In the upper layers, the deterministic NC and cooperation are proposed to improve the transmission on the packet level.

Our expectation says that the outage probability is minimized and the efficiency is maximized. Moreover, the optimal solutions could be investigated to determine the best channel coding rate with best deterministic NC and cooperative protocols and then compared with the random NC or any other techniques.

Other direction of the future work is applying the deterministic NC and cooperation over LTE-A with video transmission to compare it with the unequal error correction and the random linear codes and random NC, as it is believed that the more 4G standard evolves the more heterogeneous nodes are added to the environment, it will be more common to have a number of devices grouped within the proximity of each other. When there are a number of devices placed close to each other, this is known as a cluster. All of the devices within this cluster can cooperate with each other and with other transmitting devices, just like a normal node or device except the devices within a cluster experiences advantages they would not individually. As well as being cooperatively connected between each other, each device is also connected to the cellular network. This means the devices in a cluster will have better communication performance in terms of, bandwidth usage, end system complexity and energy efficiency. These improvements will come through a number of methods, such as sharing cellular signal between devices. There are also functional advantages. The devices contained in the cluster will provide a range of different capabilities as they will be different kinds of devices for example, computers, mobile phones, modern cars and more. These devices can then share functionality and ordinarily provide a better service throughout the cluster .This shows there are a number of advantages in developing and implementing wide, heterogeneous networks. Moreover, it is worth investigating applying deterministic NC over more practical channels than the erasure channel, such as FSMC channel.

After showing how the deterministic NC is useful in reducing the ARQ, we can go further in this direction and improve implementing NC to replacing HARQ's protocol with NC cooperative design that either improves HARQ significantly or even replace it totally.

Moreover, two different scenarios are under consideration; when all users, destination, and base stations are in range, and when just users are in range with the base stations.

References

[1] K Romer, F Mattern, and E Surich 'The design Space of Wireless Sensor Networks' IEEE wireless Communication December 2004.

[2] C. Castro , Andreas J. Kassler Carla-Fabiana Chiasserini, Claudio Casetti , and Ibrahim Korpeoglu 'Peer-to-Peer Overlay in Mobile Ad-hoc Networks Marcel' Part 9, 1045-1080, DOI: 10.1007/978-0-387-09751-0_37, 2010.

[3] T. S. Rappaport, 'Wireless Communications: principles and Practice,' 2nd Ed., Prentice-Hall: Upper Saddle River, NJ, 2002, ISBN 0-042232-0.

[4] "IEEE Standard for Information technology-Telecommunications and information exchange between systems-Local and metropolitan area networks-Specific requirements - Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications", IEEE 802.11, June 2007.

[5] ECMA International, "High Rate Ultra Wideband PHY and MAC Standard ECMA-368", 3rd Edition, December 2008, http://www.ecma-international.org /publications/files/ECMA-ST/ECMA-368.pdf.

[6] R. Ahlswede, N. Cai, S.-Y.R. Li, and .W. Yeung, "Network information flow," IEEE Trans. Inform. Theory, vol. 46, pp. 1204–1216, July 2000.

[7] E. Erkip, A. Sendonaris, A. Stefanov and B. Aazhang. 'Cooperative Communication in Wireless Systems'. In Advances in Network Information Theory, edited by Piyush Gupta, Gerhard Kramer and Adriaan J. van Wijngaarden, pp. 303-320, AMS DIMACS Series, 2004.

[8] V. Stankovic, L. Stankovic, A., Moinian, and S., Cheng, Wireless full-duplex communications based on NC, Proc. 45th Annual Allerton Conference on Communications, Control and Computing, Monticello, IL, Sept. 2007.

[9] C. Fragouli, J-Y. Boudec, and J. Widmer.: ‘Network coding: an instant primer,’ ACM Sigcomm Computer Comm. Review, 2006, vol. 36.

[10] C. Fragouli, D. Katabi, A. Markopoulou, M. Médard, and H. Rahul, “Wireless Network Coding: Opportunities Challenges,” in IEEE Military Communications Conference (MILCOM 2007), Orlando, Florida, USA, Oct. 2007.

[11] B. Li, and D. Niu, “Random Network Coding in Peer-to-Peer Networks: From Theory to Practice,” Proceedings of the IEEE , vol.99, No.3, pp.513–523, Mar. 2011.

[12] L Jingyang, W Fengwei and T Zhang ‘Network-Coded Cooperative Diversity in Multi-Hop Cellular Networks’

[13] M. Pedersen and F. Fitzek ‘Implementation and Performance Evaluation of Network Coding for Cooperative Mobile Devices’. Communications Workshops, ICC Workshops '08. IEEE International Conference on (pp. 91 - 96). Aalborg: IEEE.

[14] M. Panasonic "Proposal of bit mapping for type-III HARQ". TSG-RAN Working Group 1 Meeting #18. Boston. 56.

[15] M. Stambaugh ‘HARQ Process Boosts LTE Communications’. Retrieved Aug. 15, 2011, from Mobile Dev & Design: http://mobiledevdesign.com/standards_regulations/HARQ_Process_Boosts_LTE_Communications/index1.html.

[16] P. Frenger, S. Parkvall, and E. Dahlman "Performance comparison of HARQ with Chase combining and incremental redundancy for HSDPA". 2001 Stockholm: Ericsson Radio Syst.

[17] W. Yafeng, Z Lei and Y. Dacheng "Performance Analysis of Type III HARQ With Turbo Codes". Vehicular Technology Conference, 2003. VTC 2003-Spring. The 57th IEEE Semiannual (pp. 2740 - 2744 vol.4). Jeju: Wireless Commun. Res. Center, Beijing Univ. of Posts & Telecommun., China .

[18] P. Frenger, S. Parkvall and E. Dahlman "Performance comparison of HARQ with Chase combining and incremental redundancy for HSDPA". Vehicular Technology Conference, 2001. VTC 2001 Fall. IEEE VTS 54th. 3. Piscataway Township, New Jersey: IEEE Operations Center. pp. 1829–1833. doi:10.1109/VTC.2001.956516. ISBN 0-7803-7005-8.

[19] El. Bahri, W. Boujemaa, S. Hatem, and M. Mohammed "Comparison of Type I, II and III Hybrid ARQ Schemes Over AWGN Channel". Ariana: IEEE International Conference on Industrial Technology 2004.

[20] A. Viterbi, A. Viterbi, and N. Sindhushayan "Interleaved concatenated codes: New perspectives on approaching the Shannon limit" Proc. Natl. Acad. Sci. USA vol 94, pp 9525-9531, September 1997

[21] D Costello, J. Hagenauer, H. Imai, and S. Wicker "Applications of Error-Control Coding" IEEE, Transactions on Information Theory vol. 44, No. 6, October 1998 pp2531-2560.

[22] C. Shannon "A mathematical theory of communication," Bell Syst. Tech. J., vol. 27, pp. 379–423, 1948.

[23] R. Hamming "Error detecting and error correcting codes," Bell Syst. Tech. J., vol. 29, pp. 147–150, 1950.

[24] E. Berlekamp "The technology of error correction codes" Proc. IEEE vol. 68, Sept. 1980 564–593.

[25] C. Berrou A. Glavieux and P. Thitimajshima 'Near Shannon Limit Error-Correcting Coding and Decoding: Turbo Codes' Proc. 1993 IEEE Int. Conf. Communications, Geneva, Switzerland, pp. 1064-1070, May 1993.

[26] S. Chung G. Forney, T Richardson and R. Urbanke 'On the Design of Low Density Parity Check Codes within 0.0045 dB of the Shannon Limit' IEEE Commun. Lett., vol. 5, no.2, Feb. 2001.

[27] F. Kschischang, B. Frey and H. Loeliger 'Factor Graphs and the Sum-Product Algorithm' IEEE Trans. on Inform. Theory, vol 47, pp. 498-519, Feb. 2001.

[28] F. Kschischang, and B. Frey 'Iterative Decoding of Compound Codes by Probability Propagation in Graphical Models' IEEE Journal on Selected Areas in Commun., vol. 16, 1, Jan. 1998.

[29] R. Gallager 'Low Density Parity Check Codes' IRE Transactions on Information Theory, IT-8, pp. 21-28, January 1962.

[30] R. Gallager, 'Low Density Parity Check Codes' MIT Press, Cambridge, Mass., 1963.

[31] D. MacKay, and R. Neal 'Near Shannon Limit Performance of Low Density Parity Check Codes' Electronics Letters vol. 32, no. 18, pp. 1645-1646, 1996.

[32] R. Tanner 'A Recursive Approach to Low Complexity Codes' IEEE Trans. on Inform Theory, vol. IT-27, pp. 533-547, Sept. 1981.

[33] Y. Kou, S. Lin, and M. Fossorier 'Construction of Low Density Parity Check Codes: A Geometric Approach' Proc. 2nd Int. Sypm. Turbo Codes and Related Topics, pp. 137-140, Brest, France, Sept. 4-7, 2000.

[34] N. Deo, 'Graph Theory with Applications to engineering and Computer Sciences' Prentice Hall, Englewood, NJ, 1974.

[35] J. Massey 'Threshold Decoding' MIT Press, Cambridge, Mass. 1963.

[36] S. Lin and D. Costello 'Error Control Coding: Fundamentals and Applications' Prentice Hall, Englewood Cliffs, NJ, 1983.

[37] Y. Kou, S. Lin, S. and M. Fossorier 'Low Density Parity Check Codes Based on Finite Geometries: A Discovery and New Results' IEEE Trans. on Inform. Theory vol IT-47, Nov. 2001.

[38] D. MacKay, 'Good Error-Correcting Codes based on Very Sparse Matrices' IEEE Trans. on Inform. Theory, vol IT-45, pp. 399-432, Mar. 1999.

[39] L. Fagoonee 'A Multi-functional Turbo Receiver based on Partial Unit Memory Codes". PhD Thesis Lancaster university. May 2003.

[40] G. Lauer "Some optimal partial unit-memory codes", IEEE Transactions on Information Theory, IT-25, pp. 240–243, March 1979.

[41] C. Nelson, L. Stankovic, V. Stankovic and S. Cheng, "Performance Analysis of PUM codes", Proc. 5th International Symposium on Turbo Codes & Related Topics, Lausanne, Switzerland, September 2008.

[42] C. Nelson, L Stankovic, V. Stankovic, and S. Cheng 'Performance Analysis of PUM codes' Proc. 5th International Symposium on Turbo Codes & Related Topics, Lausanne, Switzerland, September 2008.

[43] R. Ahlswede, N. Cai, S. Li, and R. Yeung, “Network information flow,” IEEE Trans. Inform. Theory, vol. 46, no. 4, pp. 12041216, 2000.
[44] J. Jin, B. Li, and T. Kong, “Is random network coding helpful in wimax?,” in IEEE INFOCOM 2008, Phoenix, AZ, April 2008.
[45] J. Byers, L. Michael, M Mitzenmachert, and A. Rege ‘A Digital Fountain Approach to Reliable Distribution of Bulk Data’ ACM SIGCOMM Computer Communication, Volume 28, Issue 2, ACM New York, NY, USA Oct. 1998
[46] ‘ETSI TR 102 591’, digital video broadcast (dvb); ip data cast; Content delivery protocols (cdp) implementation guidelines part 1: IP data cast over dvb,’ ETSI Tech. Spec., 2007.
[47] M. Lnby, M. Mitzenmacher, A. Shokrollahi, D. Spielman, and V. Stemann, “Practical Loss-Resilient Codes.” In Proceedings of the 2gth ACM Symposium on Theory of omputing, 1997.
[48] A. Shokrollahi, “Raptor codes,” IEEE Trans. Inform. Theory, vol. 52, no. 6, pp. 2251–2567, 2006.
[49] “Etsi ts 126 346, universal mobile telecommunications system (umts); multimedia broadcast/multicast service (mbms); protocols and codecs,” ETSI Tech. Spec., 2005.
[50] T. Stockhammer, A. Shokrollahi, M. Watson, M. Luby, and T. Gasiba, Application Layer FEC for Mobile Multimedia Broadcasting (in Handbook of Mobile Broadcasting), CRC Press (Eds: B. Furht, and S. Ahson), 2008.
[51] HE Hao Singapore Institute of Manufacturing Technology, “Some Applications of Wireless Sensor Networks in SIMTech” 23 Jan 07.
[52] Website: ‘http://www.3gpp.org/LTE-Advanced’, retrieved date Aug. 2011.
[53] Website ‘http://3g.harkul.com/a-quick-primer-on-coordinated-multi-point-comp-technology/’ access date 23rd Sept 2011.
[54] NTT Docomo technical journal vol.12 No. September 2010.
[55] M. Katz ‘Cooperative Networking with Wireless Device Technologies for Building Future Wireless Grids’. Celtic Information Day (Call 6) (pp. 3-15). Helsinki, Finland: VTT.

[56] G. Youjun, W. Qixing and L. Guangyi ‘The Access Network and Protocol Design’. In Wireless Communication Networking and Mobile Computing (pp. 1-4). Beijing: China Mobile Research Institute (CMRI), China Mobile Communications Corporation (CMCC).

[57] T. Ho, M. M´edard, R. Koetter, D. Karger, M. Effros, J. Shi, and B. Leong, “A random linear network coding approach to multicast,” IEEE Transcactions on Information Theory, vol. 52, no. 10, pp. 4413–4430, Oct. 2006.

[58] Z. Ghadialy ‘Coordinated Multi-Point (CoMP) transmission and reception.’. Retrieved November 7, 2010, from 3G and 4G Wireless Blog: http://3g4g.blogspot.com/2010/02/coordinated-multi-point-comp.html.

[59] S Parkvall ‘LTE-Advanced – Evolving LTE towards IMT-Advanced’. (pp. 1-5). Stockholm: Ericsson Research.

[60] D. Nguyen, T. Tran, T. Nguyen, and B. Bose ‘Hybrid ARQ-Random Network Coding for Wireless Media Streaming’. Communications and Electronics, 2008. ICCE 2008.

[61] S. Hong, and J. Chung ‘Network-coding-based hybrid ARQ scheme for mobile relay networks". Electronics Letters (pp. 539 -541).

[62] Y. Sun, Y. Li, and X. Wang ‘Cooperative Hybrid-ARQ Protocol with network Coding’. Communications and Networking in China, 2009.

[63] M. Sawahashi, Y. Kishiyama, A. Morimoto, D. Nishikawa, and M. Tanno ‘Coordinated multipoint transmission/reception techniques for LTE-advanced [Coordinated and Distributed MIMO]’. Wireless Communications, IEEE (pp. 26-34). Tokyo: IEEE Communications Society.

[64] 3GPP. (2010). "LTE-Advanced". Retrieved October 18, 2010, from 3GPP - A Global Initiative: http://www.3gpp.org/LTE-Advanced.

[65] Website ‘http://www.freescale.com/’ retrieving date March 2011.

[66] R. Berlekamp ‘Algebraic Coding theory’. Revised 1984 edition. Aegean Park Press 1984.

[67] S. Lin and D. Costello, ‘Error Control Coding: Fundamentals and Applications’ Prentice Hall, Englewood Cliffs, NJ, 1983.

[68] F. MacWilliams, and N. Sloane. 'The Theory of Error-Correcting Codes. Elsevier Science', 1977.

[69] J. McEliece, 'The Theory of Information and Coding', second edition. Cambridge University Press 2002.

[70] W. Peterson, and E. Weldon 'Error Correcting Codes' MIT Press, Cambridge, MA. 1972.

[71] F. Hiromasa, S Takahiko and S. IWAO "Turbo-codes decoding with weighting of extrinsic information for punctured turbo-codes" IEIC Technical Report (Institute of Electronics, Information and Communication Engineers), ISSN:0913-5685, Japan, 2000.

[72] J. Nicholas, A. Harvey, R. David, K. Karger, and K. Murotay 'Deterministic Network Coding by Matrix Completion' SODA 2005: 489-498.

[73] E. Casas, and C. Leung 'A Simple Digital Fading Simulator for Mobile Radio' IEEE Transactions on Vehicular Technology, 39(3).

[74] M. Médard, and D. Katabi, 'Introduction to Network Coding'. Retrieved December 5, 2010, from MIT Short Programs: http://web.mit.edu/professional/short-programs/courses/network_coding.html

[75] K. Javed "ZigBee suitability for Wireless Sensor Networks in Logistic Telemetry Applications" Halmstad University, 2006.

[76] K. Vivek "Multiple Description Coding: Compression Meets The Network," IEEE Signal Processing Mag., vol. 8, no. 5, pp. 74–93, Sept. 2001.

[77] M. Wang and B. Li, "How practical is network coding?," IEEE IWQoS 2006, Intl. Workshop on Quality of Service, pp. 274-278.

[78] S. Nazir, D. Vukobratović, and V. Stanković , "Performance evaluation of Raptor and Random Linear Codes for H.264/AVC video transmission over DVB-H networks," in Proc. IEEE ICASSP-2011, Intl. Conf. on Acoustics, Speech, and Signal Processing, Prague, Czech Republic, May 2011.

[79] T. Tillo, M. Grangetto, and G. Olmo, "Redundant slice optimal allocation for H.264 multiple description coding," IEEE Trans. On Circuits and Systems for Video Technology, vol. 18, no. 1, Jan 2008.

[80] S. Mao, S. Lin, S. Panwar, Y. Wang, and E. Celebi, "Video transport over ad hoc networks: Multistream coding with multipath transport," IEEE J. on Selected Areas in Communs., vol. 21, no. 10, pp. 1721–1737, 2003.

[81] Glats, P., Hein, K., and Weiss, R.: „Energy Conservation with Network Coding For Wireless Sensor Networks with Multiple Crossed Information Flows,‟ Proc. 10th International Symposium on Pervasive, Algorithms, and Network, Taiwan, December, 2009, pp 201-207.

[82] Liang, X., Chen, M., Xiao, Y., Balasingham, I., and Leung, V.: „A Novel Cooperative Communication Protocol for QoS Provisioning in Wireless Sensor Network,‟ Proc. 5th Int. Conf. on Testbeds and Research Infrastructures for the Development of Networks & Communities and Workshops,, Washington, April, 2009.

[83] V. Stankovic, Z. Xiong, and A. Host-Madsen, "Cooperative Diversity for Wireless Ad Hoc Networks: Capacity Bounds and Code Designs," IEEE Signal Processing Magazine, Aug. 2006.

[84] D. Woldegebreal, and H. Karl 'Network-coding-Based Cooperative Transmission in Wireless Sensor Networks: Diversity-Multiplexing Tradeoff and Coverage Area Extension' Springer- Verlag Berlin, 2008, 4913, pp. 141-155.

[85] S. Jayaweera 'Virtual MIMO-based cooperative communication for energy-constrained wireless sensor networks' IEEE Trans. Wireless Commun., 2006, (5), pp. 984-989.

[86] S. Jayaweera, 'Energy Analysis of MIMO Techniques in Wireless Sensor Networks' Proc. 38th Annual Conf. on Information Sciences and Systems (CISS 04), Princeton, NJ, March, 2004.

[87] L. Fagoonee, and B. Honary 'Construction of partial unit memory encoders for application in capacity-approaching concatenated codes' IEE Proc. Comm.: Special Issue on Capacity Approaching Codes, Design and Implementation, 2005, 152, (6), pp. 1108-1115.

[88] C. Nelson, L. Stankovic, L., and B. Honary 'Partial-Unit Memory based Turbo Codes' IET Electronics Letters, 2009, 45, (21).

[89] C. Hausl, F. Schreckenbach, I. Oikonomidis, and G. Bauch 'Iterative network and channel decoding on a Tanner graph' Proc. Allerton, Monticello, IL, Sept. 2004.

[90] C. Hausl, and J. Hagenauer 'Iterative network and channel decoding for the two-way relay channel' Proc. IEEE ICC, Istanbul, Turkey, June 2006.

[91] L. Xiao, T. Fuja, J. Kliwer, and D. Costello 'Nested Codes with Multiple Interpretations' Proc. CISS, Princeton, USA, March, 2006.

[92] P. Popovski, and H. Yomo 'The Ant-Packets Can Increase the Achievable Throughput of a Wireless Multi-Hop Network' Proc. IEEE ICC, Istanbul, Turkey, June 2006.

[93] P. Popovski, and H. Yomo 'Bi-directional amplification of throughput in a wireless multi-hope network,' in Proc. IEEE 63rd VTC, Melbourne, Australia, May 2006.

[94] H. Gacanin and F. Adachi, "Broadband analog network coding," IEEE Trans. on Wireless Communications, Vol. 9, No. 5, pp. 1577-1583, May 2010.

[95] L. Jianquan, T. Meixia, X. Youyun, and W. Xiaodong 'Superimposed XOR: 'A New Physical Layer Network Coding Scheme for Two-Way Relay Channels' . IEEE Communications Society subject matter experts for publication in the IEEE "GLOBECOM" proceedings, 2009.

[96] C. Aitor and I. Christian 'Achievable Rates for the AWGN Channel with Multiple Parallel Relays' IEEE Trans. Wireless Communications, Vol. 8, Issue 5, May 2009

[97] A. El-Gamal, M. Mohseni and S. Zahedi, 'Bounds on Capacity and Minimum Energy-Per-Bit for AWGN Relay Channels Fellow', IEEE Tras. On Information Theory, Vol. 52, No. 4, April 2006

[98] S. Katti, I. Maric, D. Katabi, A. Goldsmith, and M. Medard: 'Joint relaying and network coding in wireless networks,' Proc. IEEE ISIT, Nice, France, June, 2007.

[99] T Cover and A. Gamal, "Capacity theorems for the relay channel," IEEE Trans. Inform. Theory, vol. 25, pp. 572–584, Sep. 1979.

[100] J. Laneman and G. Wornell, "Energy-efficient antenna sharing and relaying for wireless networks," IEEE Wireless Comm. And Network Conference, Sep 2000.

[101] J. Laneman, D. Tse and G. Wornell, “Cooperative diversity in wireless networks: Efficient protocols and outage behaviour,” IEEE Trans. Inform. Theory, vol. 50, no. 12, pp. 3062–3080, Dec. 2004.

[102] A. Stefanov and E. Erkip, “Cooperative coding for wireless networks,” IEEE Trans. Commun., vol. 52, Sept. 2004.

[103] T. Hunter and A. Nosratinia, “Diversity through coded cooperation,” IEEE Trans. Wireless Commun., vol. 5, Feb. 2006.

[104] K. Azarian, H. El-Gamal, and P. Schniter, “On the Achievable Diversity-Multiplexing Tradeoff in Half-Duplex Cooperative Channel,” IEEE Trans.Inform. Theory, vol. 51, no. 12, Dec 2005.

[105] L. Xiao, T. Fuja, J. Kliewer and D. Costello “A network coding approach to cooperative diversity,” IEEE Trans. Inform. Theory, vol. 53, no. 10, Oct. 2007.

[106] M. Yu, J. Li, and R. Blum, “User cooperation through network coding,” IEEE Int’l Conf. Comm., Glasgow, U.K., June 2007.

[107] S. Y.R.Li, R. Yeung and N. Cai, “Linear network coding,” IEEE Trans. Inform. Theory, vol. 49, no. 2, pp. 371-381, Feb. 2003.

[108] J. van Lint and R. Wilson. A Course in Combinatorics, Cambridge University Press, 1992.

[109] S. Katti, H. Rahul, W. Hu, D. Katabi, M. Medard, and J. Crowcroft, “XORs in the Air: Practical Wireless Network Coding,” Proc ACM SIGCOMM, 2006.

[110] Q. Zhang, and S. Kassam, “Finite-State Markov Model for Rayleigh Fading Channels,” IEEE Transactions on Communications, Vol. 47, No.11, pp. 1688-1692, November 1999.

[111] T. Rappaport, “Wireless Communications’,’ Prentice Hall, Upper Saddle River, NJ, 2001.

[112] D. Puccinelli and M. Haenggi, “Multipath Fading in Wireless Sensor Networks: Measurements and Interpretation,” Proc. ACM IWCMC 2006, Vancouver, Canada, July 2006.

[113] E. Magli and P. Frossard: ‘An overview of Netwrok Coding for multimedia streaming’ Multimedia and Expo, pages 1488-1491, ICME 2009.

[114] M. Wu, S. Karande, and H. Radha, “Network Embedded FEC for Optimum Throughput of Multicast Packet Video,” EURASIP Journal on Advances in Signal Processing, vol.20, no. 8, pp. 728–742, Sep. 2005.

[115] C. Wu and B. Li, “Optimal Peer Selection for Minimum- Delay Peer-to-Peer Streaming with Rateless Codes,” in Proc. of ACM P2PMMS, Singapore, Nov. 2005, pp. 69–78.

[116] N. Thomos and P. Frossard, “Raptor Network Video Coding,” in Proceedings of the 1st ACM International Workshop on Mobile video (in conjunction with ACM Multimedia 2007), Augsburg, Germany, Sep. 2007.

[117] Advances in Device-to-Device Communications and Network Coding for IMT-Advanced

[118] E. and P. Frossard ‘An overview of network coding for multimedia streaming’ Multimedia and Expo, ICM 2009.

[119]L. Yuxing. ‘Network Coding for Peer-to-Peer Live Media Streaming’ Grid and cooperative commuting, fifth international conference, pages 149-155 GCC Oct. 2006.

[120] H. Seferoglu, H. and A. Markopoulou, ‘Opportunistic Network Coding for Video Streaming over Wireless’ Packet video, pages 191-200. Dec. 2007.

[121] I. Glover, and P. Grant, M. Digital Communications. Harlow, England : Prentice Hall, 2010. pp. 471-477.

[122] 3GPP. Further Advancements for EUTRA Physical Layer Aspects. Mar. 2010. TR 36.814 V9.0.0.

[123] A Agrawal, ‘Heterogeneous Networks A new paradigm for increasing cellular capacity’ Sr VP, Technology, Qualcomm, Jon, 2009.