Towards Reliable Communication in LTE-A Connected Heterogeneous Machine to Machine Network

by

Ahasanun Nessa

THESIS PRESENTED TO ÉCOLE DE TECHNOLOGIE SUPÉRIEURE
IN PARTIAL FULFILLMENT FOR THE DEGREE OF
DOCTOR OF PHILOSOPHY
Ph.D.

MONTREAL, JUNE 19, 2017

ÉCOLE DE TECHNOLOGIE SUPÉRIEURE
UNIVERSITÉ DU QUÉBEC

BOARD OF EXAMINERS

THIS THESIS HAS BEEN EVALUATED

BY THE FOLLOWING BOARD OF EXAMINERS

M. Michel Kadoch, Thesis Supervisor
Department of Electrical Engineering, École de Technologie Supérieure

M. Bo Rong, Co-supervisor
Communications Research Center Canada, Ottawa, ON K2H 8S2,Canada

M. Jean-Marc Robert, President of the Board of Examiners
Department of Software Engineering & IT, École de Technologie Supérieure

M. François Gagnon, Member of the jury
Department of Electrical Engineering, École de Technologie Supérieure

M. Walaa Hamouda, External Independent Examiner
Department of Electrical & Computer Engineering, Concordia University

THIS THESIS WAS PRESENTED AND DEFENDED

IN THE PRESENCE OF A BOARD OF EXAMINERS AND THE PUBLIC

ON APRIL 20, 2017

AT ÉCOLE DE TECHNOLOGIE SUPÉRIEURE

ACKNOWLEDGEMENTS

I wish to express my sincere thanks to my supervisor, Prof. Michel Kadoch for his constant support, valuable advices and encouragements during the course of this work. He was always available for my questions and was very positive about my research work. My special thanks go out to my co-supervisor Bo Rong, who showed me the road and helped me a lot along the path of this degree. His understanding, guidance and both detailed and constructive comments have provided a good basis for the present thesis and my graduate research career. Working with both of them have added a valuable experience to my professional life.

I would also like to thank other members of my thesis committee: Prof. Jean-Marc Robert at Department of Information Technology, École de technologie supérieure, and Prof. François Gagnon at Department of Electrical Engineering, École de technologie supérieure and Prof. Walaa Hamouda at Department of Electrical & Computer Engineering, Concordia University for their valuable feedback on the thesis.

My special appreciation goes to my parents and relatives for their endless love, support, encouragement, and prayers. I am forever indebted to my beloved husband Abdullah Al Kafi and my lovely son Ayman for their understanding, endless patience and encouragement when it was most required. I would also like to thank all my friends in Montreal for their help when I arrived here and during my doctoral studies.

Finally, thanks to all my colleagues of the LAGRIT Lab and other staffs of École de Technologie Supérieur.

VERS DES COMMUNICATIONS FIABLES DANS LE RÉSEAU HÉTÉROGÈNE MACHINE À MACHINE CONNECTÉE AU LTE-A

Ahasanun Nessa

RÉSUMÉ

La communication machine à machine (M2M) est une technologie émergente qui permet à des dispositifs hétérogènes de communiquer entre eux sans intervention humaine et donc de former le soi-disant Internet des choses (IoTs). Les réseaux cellulaires sans fil (WCN) jouent un rôle important dans le déploiement réussi de la communication M2M. En particulier, le déploiement massif continu de l'évolution à long terme (LTE-A) permet d'installer le réseau M2M dans la plupart des zones urbaines et éloignées, et en utilisant le réseau backhaul LTE-A, une communication en réseau sans faille est établie entre les nœuds MTC et -applications. Cependant, la couverture étendue du réseau ne garantit pas une mise en œuvre réussie de la communication M2M dans le LTE-A, et il ya donc encore des défis.

Une transmission fiable et efficace en l'énergie est peut-être la demande des plus importante pour diverses applications M2M. Parmi les facteurs qui affectent la fiabilité de la communication M2M, figurent le retard élevé de bout en bout et le taux élevé d'erreur binaire. L'objectif de la thèse est de fournir une communication M2M fiable dans le réseau LTE-A. Dans ce but, pour alléger la congestion de signalisation sur l'interface aérienne et l'agrégation de données efficace, nous considérons une architecture basée sur le cluster où les périphériques MTC sont regroupés en nombre de clusters et les trafics sont transmis à travers certains nœuds spéciaux appelés tête de cluster (CHs) aux Stations de Base (BS) en utilisant des transmissions mono ou multisaut. Dans de nombreux scénarios de déploiement, certaines machines sont autorisées à se déplacer et à changer leur emplacement dans la zone de déploiement à très faible mobilité. En pratique, les performances de la transmission de données se dégradent souvent avec l'augmentation de la distance entre CHs voisins. CH doit être réexaminé dans de tels cas. Cependant, la réélection fréquente des CH entraîne un contre-effet sur le routage et la reconfiguration de l'allocation des ressources associée aux protocoles qui dépendent du CH. En outre, la qualité de la liaison entre un CH-CH et un CH-BS est très souvent affectée par divers facteurs environnementaux dynamiques tels que la chaleur et l'humidité, les obstacles et les interférences RF. De nombreuses solutions ont été proposées pour lutter contre le canal sans fil à risque d'erreur, comme la demande de répétition automatique (ARQ) et le routage multipath. Dans le premier schéma, l'émetteur retransmet le paquet entier même si la partie du paquet a été reçue correctement et dans le dernier, le récepteur peut recevoir les mêmes informations à partir de chemins multiples ; donc les deux techniques sont inefficaces en rapport à la bande passante et l'énergie. En outre, avec la retransmission, le délai global de bout en bout peut dépasser le budget de délai maximal autorisé.

Sur la base des observations susmentionnées, nous identifions le canal CH à CH comme étant l'un des goulots d'étranglement pour fournir une communication fiable dans le cluster basé sur le réseau hiérarchique M2M et nous présentons une solution complète pour soutenir les communications coopératives codées par fontaine. Notre solution couvre de nombreux aspects, de la sélection de relais à la formation coopérative pour répondre aux exigences de QoS de l'utilisateur. Dans la première partie de la thèse, nous concevons d'abord un algorithme de sélection de relais incrémentiels codés sans raillerie (RCIRS) basé sur des techniques gourmandes pour garantir le débit de données requis avec un coût minimum. Ensuite, nous développons des protocoles de communication coopératifs codés par fontaine pour faciliter la transmission de données entre deux CH voisins. Dans la deuxième partie, nous proposons des systèmes de codage de réseau commun et de fontaines pour une communication fiable. Grâce au codage de canal de couplage et au codage de réseau, simultanément dans la couche physique, les schémas de codage de réseau conjoint et de fontaine exploitent efficacement la redondance des deux codes et combattent efficacement l'effet néfaste des conditions d'affaiblissement du signal dans des canaux sans fil. Dans le schéma proposé, après décodage correct des informations provenant de sources différentes, un noeud relais applique un codage réseau et fontaine sur les signaux reçus puis transmet à la destination dans une seule transmission. Par conséquent, les schémas proposés exploitent la diversité spatiale et améliorent l'efficacité de la bande passante. Dans la troisième partie, nous nous concentrons sur la transmission uplink fiable entre CHs et BS où les CHs transmettent directement à BS ou à l'aide des nœuds de relais LTE-A (RN). Nous étudions le réseau LTE-A amélioré de type I et de type II et proposons un ensemble conjoint de systèmes de codage réseau et fontaine qui permettent de lutter efficacement contre l'effet néfaste des canaux d'évanouissement sans fil et d'améliorer la robustesse des liaisons.

Enfin, les solutions proposées sont évaluées par des simulations numériques étendues et les résultats numériques sont présentés pour fournir une comparaison avec les travaux connexes trouvés dans la littérature.

Mots clés: Internet of Things (IoT), Machine to machine (M2M) communication, Cooperative Communications, Fountain Codes, Network code.

TOWARDS RELIABLE COMMUNICATION IN LTE-A CONNECTED HETEROGENEOUS MACHINE TO MACHINE NETWORK

Ahasanun Nessa

ABSTRACT

Machine to machine (M2M) communication is an emerging technology that enables heterogeneous devices to communicate with each other without human intervention and thus forming so-called Internet of Things (IoTs). Wireless cellular networks (WCNs) play a significant role in the successful deployment of M2M communication. Specially the ongoing massive deployment of long term evolution advanced (LTE-A) makes it possible to establish machine type communication (MTC) in most urban and remote areas, and by using LTE-A backhaul network, a seamless network communication is being established between MTC-devices and -applications. However, the extensive network coverage does not ensure a successful implementation of M2M communication in the LTE-A, and therefore there are still some challenges.

Energy efficient reliable transmission is perhaps the most compelling demand for various M2M applications. Among the factors affecting reliability of M2M communication are the high end-to-end delay and high bit error rate. The objective of the thesis is to provide reliable M2M communication in LTE-A network. In this aim, to alleviate the signalling congestion on air interface and efficient data aggregation we consider a cluster based architecture where the MTC devices are grouped into number of clusters and traffics are forwarded through some special nodes called cluster heads (CHs) to the base station (BS) using single or multi-hop transmissions. In many deployment scenarios, some machines are allowed to move and change their location in the deployment area with very low mobility. In practice, the performance of data transmission often degrades with the increase of distance between neighboring CHs. CH needs to be reselected in such cases. However, frequent re-selection of CHs results in counter effect on routing and reconfiguration of resource allocation associated with CH-dependent protocols. In addition, the link quality between a CH-CH and CH-BS are very often affected by various dynamic environmental factors such as heat and humidity, obstacles and RF interferences. Since CH aggregates the traffic from all cluster members, failure of the CH means that the full cluster will fail. Many solutions have been proposed to combat with error prone wireless channel such as automatic repeat request (ARQ) and multipath routing. Though the above mentioned techniques improve the communication reliability but intervene the communication efficiency. In the former scheme, the transmitter retransmits the whole packet even though the part of the packet has been received correctly and in the later one, the receiver may receive the same information from multiple paths; thus both techniques are bandwidth and energy inefficient. In addition, with retransmission, overall end to end delay may exceed the maximum allowable delay budget.

Based on the aforementioned observations, we identify CH-to-CH channel is one of the bottlenecks to provide reliable communication in cluster based multihop M2M network and present a full solution to support fountain coded cooperative communications. Our solution covers many aspects from relay selection to cooperative formation to meet the user's QoS requirements. In the first part of the thesis, we first design a rateless-coded-incremental-relay selection (RCIRS) algorithm based on greedy techniques to guarantee the required data rate with a minimum cost. After that, we develop fountain coded cooperative communication protocols to facilitate the data transmission between two neighbor CHs. In the second part, we propose joint network and fountain coding schemes for reliable communication. Through coupling channel coding and network coding simultaneously in the physical layer, joint network and fountain coding schemes efficiently exploit the redundancy of both codes and effectively combat the detrimental effect of fading conditions in wireless channels. In the proposed scheme, after correctly decoding the information from different sources, a relay node applies network and fountain coding on the received signals and then transmits to the destination in a single transmission. Therefore, the proposed schemes exploit the diversity and coding gain to improve the system performance. In the third part, we focus on the reliable uplink transmission between CHs and BS where CHs transmit to BS directly or with the help of the LTE-A relay nodes (RN). We investigate both type-I and type-II enhanced LTE-A networks and propose a set of joint network and fountain coding schemes to enhance the link robustness.

Finally, the proposed solutions are evaluated through extensive numerical simulations and the numerical results are presented to provide a comparison with the related works found in the literature.

Keywords: Internet of things (IoT), machine type communications(MTC), cooperative communications, fountain codes, network coding

TABLE OF CONTENTS

Page

LIST OF TABLES

LIST OF FIGURES

LIST OF ABREVIATIONS

3GPP	Third Generation Partnership Project
AGTI	Access grant time interval
ARQ	Automatic Repeat Request
BFS	Brute-Force Search
CHs	Cluster Heads
CSI	Channel Sate Information
E-UTRA	Evolved Universal Terrestrial Radio Access
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
eNB	LTE-A eNodeB
EPC	Evolved Packet Core
EPS	Evolved Packet System
E-UTRAN	Evolved Universal Terrestrial Radio Access Network
E-UTRA	Evolved Universal Terrestrial Radio Access
FDD	Frequency Division Duplex
H2H	Human-to-Human
HARQ	Hybrid Automatic Repeat Request
HSPA	High Speed Packet Access
HSS	Home Subscriber Server
ILLA	Inner Loop Link Adaptation

IMT	International Mobile Telecommunications
IoT	Internet of Thing
ITU	International Telecommunication Union
ITU-R	ITU Radiocommunication Sector
JNFC-RLNC	JNFC with Random Linear Network Coding
JNFC-RSD	JNFC with Robust Soliton Distribution
LTE	Long Term Evolution
LTE-A	Long Term Evolution Advanced
M2M	Machine to Machine
M-BUS	Meter Bus
MAC	Medium Access Control.
MCNs	Mobile Cellular Networks
MBSFN	Multicast Broadcast Single Frequency Network
MCS	Modulation and Coding Schmes
MIMO	Multiple Input Multiple Output
MTC	Machine Type Communication
OFDM	Frequency-division Multiplexing item [OLLA] Outer Loop Link Adaptation
PDNGW	Packet Data Network Gateway
PRACH	Physical Rrandom Access Channel
RA	Random Access (RA)

RCIRS	Rateless Coded Iincremental Relay Selection
RLNC	Random Linear Network Coding
RN	Relay Nodes
RV	Redundancy Version
SAE	System Architecture Evolution
SAW	stop and Wait
SGW	Serving Gateway
SNFC	separated Network and Fountain Coding
SRD	Short-Range Device
STBC	space-Time Block Coding
TDD	Time Division Duplex
TD-SCDMA	Time Division Synchronous Code Division Multiple Access
TTI	Transmission Time Interval
UWB	Ultra Wide Band
V-MIMO	Virtual MIMO
WSNs	Wireless Sensor Networks

INTRODUCTION

Day by day innovative technologies are being introduced in the communication industry to enhance people's life through facilitating the interaction between human and the rest of the world. Among them Internet of Thing (IoT) has received considerable attention from both academia and industry. In IoT real life objects are equipped with either sensors or actuators and connected with each other to transfer their sensed data to centralized servers, where information is collectively stored and made available for particular users with proper access rights. Smart infrastructure, healthcare, supply chains/logistics, and social applications are a few possible examples in which this new paradigm may play a leading role to improve people's life.

In fact wireless sensor network (WSN) is one of the most important elements in the IoT paradigm. As an important data resource in IoT, WSN collects the information from the surrounding environments, processes the collected data and transmit to the sink node. With the emergence of IoT, the traditional sensor nodes are being replaced by more technologically advanced nodes, namely machine nodes those have higher information processing features. This introduces a new paradigm in communication research, namely machine type communication (MTC) or machine to machine (M2M) communication where devices communicate with each other autonomously. Since it does not need direct human intervention, many promising real-time monitoring applications including e-healthcare, smart homes, environmental monitoring, and industrial automation can be revolutionized by this emerging technology. Similar to the traditional wireless sensor nodes, machine type communications (MTC) devices have a simple design and transmit with a low power to conserve energy and are very often installed in a variety of application environments where hundreds or thousands of machine type devices (nodes) are densely located over small or medium area (Y. Zhang & Gjessing, 2012).

In the past, tremendous technology development and commercial success have incurred in mobile cellular networks (MCNs). The well established infrastructures, wide area coverage and high-performance capabilities of cellular networks make it possible to install the M2M network in remote areas in a fast and economic way. In cellular network paradigm, Long Term Evolution-Advanced (LTE-A) is the latest cellular communication standard that is developed by the third generation partnership project (3GPP) to meet the requirements of IMT-Advanced systems, including increasing spectral efficiency and bandwidth, improving cell edge performance and mobility support, extending coverage as well as reducing latency and handoff failures (ITU-R-M.2134, 2008). In order to improve capacity and provide ubiquitous wireless access in both indoor and outdoor, LTE-A introduces a heterogeneous network (HetNet) architecture that consists of different kinds of nodes in terms of transmission power and radio frequency (RF) coverage area. By providing ubiquitous connections among M2M devices, LTE-A can play an important role to integrate remote M2M devices within IoT platform.

Nevertheless, LTE/LTE-A is optimized for human-to-human (H2H) communications such as voice or web browsing and bandwidth demanding services that is quite different than the features and service requirements of M2M communications such as massive transmissions, small bursty natured traffic, little or no mobility, requirements of high energy efficiency and security, etc. Although in many M2M communications small sizes of data need to be transmitted, when a large number of M2M devices try to connect to an LTE-A base station (e.g. eNodeB), the network will be congested and will cause too many packet collisions that effect the quality of both M2M and H2H communications. In addition the resource limitations of end point devices, lack of central administrative control, mobility of the nodes and error prone wireless channel make the reliable transmission in M2M network more harder. Hence, it is crucial yet challenging to achieve reliability in M2M communications over LTE-A network without sacrificing the quality of the H2H communications.

0.1 Problem Overview

Challenges to achieve reliability in M2M communication over cellular network (e.g, LTE/LTE-A) mainly originated from following three main aspects:

- First sort of problems are related to the efficient access of LTE-A air interface. Usually a LTE-A eNB covers about 5000 and 35000 households in the central and urban area, respectively (3GPP-R2-102296, 2010). If we consider smart metering, each household owns at least three intelligent meters, namely electric, gas and water, and each meter works independently as an MTC device. Thus, the number of MTC devices in one cell is at least two orders of magnitude higher than that of user equipments (UEs). When an M2M device has packets to transmit, it performs random access (RA) using the physical random access channel (PRACH) during an allowable time slot, called an access grant time interval (AGTI) or RA opportunity (i.e., RA-slots). In M2M communications, generally small amounts of data need to be transmitted. Although the data size is small, when a large number of M2M devices try to communicate over the same channel, the devices should contend to access the shared radio channels that causes network overload problem and hugely impacts the radio access and core networks of the cellular system;
- Second sort of problems are related to the wireless channel. In many applications a large number of sensors like M2M devices are installed in remote harsh environment. Due to the volume of the nodes it is most often impossible to estimate the link between M2M devices and LTE enhanced node B (eNB). In addition dynamic environmental factors such as heat, humidity and rainfall, hidden terminal problems and radio frequency (RF) interferences, negatively impact the data transmission between two devices. (Marfievici *et al.*, 2013);
- Finally the constrained resources of the end point devices adds more challenges to achieve reliability. The end point devices are usually small, lightweight, inexpensive sensor nodes

with low computation capability, limited energy, limited memory storage and narrow bandwidth. Due to bandwidth and energy inefficiency automatic repeat request (ARQ), link level retransmission and multipath routing are not suitable techniques for this network to use as against error prone channel. Because they consume more bandwidth and energy, and also significantly increase the end-to-end delay.

0.2 Tackling Reliable Communication in LTE-A Connected M2M Network

Among the above mentioned three sorts of problems mentioned in Section 0.1, problem related with congestion on air interface can be significantly controlled by node grouping or clustering (Huq *et al.*, 2015; Si *et al.*, 2015; ExaltedProject, 2011). In clustering M2M devices are divided into a number of clusters and limited number of nodes are selected as cluster heads (CHs) that are only allowed to access the LTE-A network. Each CH represents a cluster, collects data from the non CH nodes of its cluster, aggregates the collected data and finally forwards the aggregated data to the eNB with or without the help of low power LTE-A relay nodes. Clustering also an effective way to simplify network management and routing protocol in a large network. However, in most existing works (Huq *et al.*, 2015; Si *et al.*, 2015) on CH based M2M network, CHs just act as a simple coordinator node that collect the data from cluster members and simply forward the collected data to the eNB. Therefore, the level of congestion on air interface increases with the increase of the number of CHs. On the other hand, accommodating a large number of nodes in a cluster may also cause congestion inside the cluster. Furthermore, in the above mentioned single hop clustering, the CHs that are farthest from the eNB always spend more energy than the CHs closer to the eNB; that results in a non-uniform energy drainage pattern in the network.

To address this problem hierarchical cluster based M2M network has been proposed in some research works (Park *et al.*, 2015; Malak *et al.*, 2016) where all the nodes including cluster heads and sink node are related with each other in a child-parent relations. In this case, the non CH nodes in each cluster send their data to the respective CH and traffics are routed through the CHs to the LTE-A eNB. However, in many deployment scenarios, some devices are allowed to move and change their location in the deployment area with very low mobility, such as soldiers in battlefield surveillance applications, animals in habitat monitoring applications, and buses in a traffic monitoring applications (Zhu *et al.*, 2011; Ekici *et al.*, 2006). As a consequence, when the neighboring CHs move apart from each other because of the mobility of the nodes, the performance of the data transmission between two neighbor CHs may deteriorate below the threshold level. Therefore, CH needs to be reselected in such cases. However, frequent reselection of CHs results in counter effect on routing and reconfiguration of resource allocation associated with CH-dependent protocols.

Due to the movement of the nodes or change of wireless transmission conditions, when a receiving CH fails to decode a packet successfully, it generates a retransmission request to its upstream source CH. To combat the link level packet loss and to increase the end-to-end transmission reliability, networks use packet retransmissions or multi-path routing . When links are weak, packet retransmissions are expensive since the energy spent on a failed node to-node transmission is completely wasted. Moreover, link-level retransmissions considerably increase the end-to-end delay and waste of network capacity. In real-time traffic, link-level retransmission may not be the right approach for increasing the end-to-end transmission reliability. On the other hand in multi-path routing the receiver may receives the same information from multiple path that is bandwidth and energy inefficient. Similarly, multi-path routing is bandwidth and energy in efficient since in this approach the receiver may receives the same information from multiple path that does not bring any benefit.

In wireless communication reliable communication can also be provided by using redundant information to recover errors in the original information, which can be added either inside a packet known as error correction or across multiple packets known as erasure correction. Channel coding is an error correction technique that alone is not sufficient to reliably transmit a message from source to destination in error prone channel. To provide extra protection erasure correction technique is applied that operates on packet level at the network layer. In erasure correction coding redundant packets are generated from the original packets and transmitted to the destination over single route or multiple routes. Network coding is one kind of erasure correction coding where intermediate nodes combine the received packets into one or multiple packets and transmit to the destination to reduce number of transmission. The newly generated packets are used at the layer above the physical layer at the receiver to recover the erroneous information. In conventional method, redundancy provided by channel coding and network coding are exploited separately at different layer. Therefore, erasure-correction decoding at upper layer cannot take advantage of the redundant information of the packets if they fail channel decoding and are discarded at the physical layer. Similarly error-correction decoding can not take advantage of the redundancy provided across the packets, since it is performed at physical layer on a single packet at a time.

In addition, various dynamic environmental factors such as heat and humidity, obstacles and RF interferences negatively affects the data transmission between CHs and BS. In physical (PHY) layer, LTE/LTE-A adopts a classical Turbo code and implement hybrid automatic repeat request (HARQ) to rectify the errors in the PHY layer. HARQ works at PHY layer in LTE-A technology but controlled by medium access control (MAC) layer. Since the code rate of Turbo code is fixed, decoding failures may occur when the channel degradation exceeds the error-correction capability of the codes specially in the time varying channel. If complete packet recovery is not possible, a negative acknowledged (NACK) is sent to the transmitter for retransmission. In severe conditions ARQ protocols require many retransmissions and have high latency, that result in packet failures and waste of radio resources over the network. In LTE-A frequency division duplex (FDD) there are 8 stop and wait (SAW) process and each

SAW process should process the data within 1 ms or 1 sub frame. Once a packet is send from a particular process, it waits for an ACK/NACK. If the received data has an error then the receiver buffers the data and requests a re-transmission from the sender. As we know that there are 8 SAW process hence each process will have to wait for 8 ms before re-transmission of the data over the air interface that implies long delay and throughput loss. Moreover, Turbo code based HARQ protocols require sophisticated channel estimation schemes in order to obtain channel state information (CSI) and to provide channel quality indicator (CQI) feedback to adapt the code rate between retransmission. In the presence of large amount of devices in the network, the transport of CQI feedback may result in a significant amount of control signaling overhead. This makes the current LTE HARQ process unsuitable for mission critical MTC applications where the end-to-end latency requirement needs to be as low as 5 ms and reliability for example packet loss rate needs to be10^{-9}.

0.3 Objective

The general objective of this research work is to contribute with practical solutions for providing reliable data transmission in LTE-A connected heterogenous M2M network. In this vain, for efficient management and congestion control, we consider a cluster based M2M network where the network is partitioned into number of clusters and traffics are forwarded through some special nodes called CHs. CHs are equipped with both a short range interface, which is used to create an ad-hoc capillary network among the devices in M2M area network, and a long range interface, through which they connect to an LTE-A network.

The first main objective is to develop different protocols for reliable intercluster communications. The performance of data transmission between two CHs very often degraded due to the distance and environmental factors. It is well known that cooperative communications is an effective way in wireless networks to combat the channel fading, improve the throughput, power efficiency and coverage (Laneman *et al.*, 2004; Zhang & Zhang, 2013; Li *et al.*, 2012). In some research work distributed beam-forming (Ochiai *et al.*, 2005b) and space-time block coding (STBC) schemes have been proposed for cooperative transmission (Dohler *et al.*, 2006; Zhou *et al.*, 2008) that require strict synchronization. However, distributed STBC and beam-forming require strict synchronization among transmitters, which is not feasible in most mobile cases. Other than previous works, to facilitate the data transmission between two neighbor CHs, we propose fountain coded cooperative communications where the source and relay nodes encode their information using fountain coding before transmit to the destination.

Indeed, in cooperative communications, relay selection plays a significant role to improve the performance of the system. In rateless coded cooperative schemes the performance is improved as more relay nodes get involved. However, employing more relay nodes results more power consumption, signalling overhead and more bandwidth usages. Therefore, which nodes and how many nodes will be selected as relays are important concerns in relay selection. In addition, finding out a desired set of nodes from a large number of relay candidates by using conventional Brute-Force Search (BFS) would require huge computational effort that limits its application in practical systems. Though comprehensive studies have been found in literature on relay selection in the cooperative communications with fixed code rate, the study on the relay selection with rateless has not received much attention. Our second main objective is to develop a novel relay selection algorithm that judicially selects minimum number of nodes between two neighboring CHs with less computational effort while meets the QoS requirements.

Another factor affecting reliable communication are high end-to-end delay and the high bit error rate. In cluster based cooperative communications, a node may play the relaying role for more than one CHs. In conventional cooperative communications, a relay is restricted to serve only a single source at a particular time. Therefore, the use of cooperative communications alone may make the system bandwidth inefficient and increase end to end delay. One way to solve this problem is to employ network coding at the relay node. Using network coding a relay node encodes the received packets from different sources into one or more outgoing packets and thus, reducing the number of transmissions and end to end delay. Channel coding is indispensable in communication to recover erroneous data. However, in the conventional method, the redundancy provided by the network coding can be used only if the lower layer delivers error free packets to the upper layer. Therefore, in the conventional methods network coding cannot utilize the redundancy provided by the channel coding. Our third objective is to fully exploit the redundancy of both codes so that the redundancy provided by network coding can be used to support the channel coding for better error protection.

The fourth main objective is to develop different protocols for reliable uplink transmission between machine type devices and BS (LTE-A eNB) with the presence of LTE-A relay nodes (RNs). In this regard, we propose several joint network and fountain coding schemes so that the redundancy provided by fountain codes and network code can be used simultaneously in the physical layer to recover the erroneous information.

0.4 Research Methodology

In this research, we address the problem of reliable data transmission over LTE-A connected heterogenous M2M networks into two phases. Our motivation is to use rateless codes instead of fixed-rate codes due to their advantages in different requirements. In both phases, we use fountain codes as channel coding techniques to avoid frequent ARQ's feedbacks and improve link reliability. In fixed-rate codes, the rate is fixed and can not adapt their rate according to the channel conditions, so there is a trade off between efficiency and reliability. To achieve best possible code rate, fixed-rate codes require channel sate information (CSI) at the transmitter which is sometimes difficult to achieve due to limited bandwidth of feedback channel. In contrast to typical fixed rate codes, in fountain coding source unconscious of CSI generates as many encoded symbols as needed by simply performing module 2 operations on the source symbols and continues its transmission until it receives an ACK from receiver. In fact, the use of fountain codes bring several advantages. Firstly, they do not require any synchronization mechanism as virtual multiple input multiple output (V-MIMO). Secondly, they naturally adapt the code rate according to the channel quality, so they do not require precise channel state information(CSI) at the transmitter as fix rate codes. Thirdly, since they naturally adapt the rate according to the channel quality, in error prone channel the transmitter does not require to increase their power to make the reception successful at the receiver.

In the first phase, we employ cooperative communications to fasciate the inter clusters communications and propose a set of schemes to support the cooperative communication between two neighboring CHs. Our solution covers many aspects from relay selection to cooperative formation to meet the user's QoS requirements. Considering the cost and performance issues in M2M network, we focus on relay selections between two neighboring CHs aiming at achieving the target data rate between two clusters with the involvement of minimum number of relay nodes. The target data rate between two clusters, which is associated with the required end to end signal quality depending on the type of applications, is provided by the network designer. When a large number of non-CH nodes are available to be a relay, finding the optimum set by using conventional Brute-Force Search (BFS) would require huge computational effort that limits its application in practical systems. In order to reduce the computational complexity of searching the optimum set, we propose rateless coded incremental relay selection (RCIRS) algorithm based on greedy technique. In our approach, source CH collects the channel state information of neighboring nodes and includes some non-CH nodes in the cooperative group one by one so that the cooperative data rate is incrementally increased until to meet the target data rate.

After that, we develop both source-feedback and non-source-feedback fountain coded cooperative communication protocols to improve reliability and robustness. We propose source-feedback protocol and non-source-feedback protocol using Raptor code (Etesami & Shokrollahi, 2006). In the source-feedback protocol, the source CH and relays transmit to the destination until the destination is capable to decode the source CH's information correctly. After successfully decoding the source CH's information, the destination CH sends acknowledgement (ACK) to source and relays. In the non-source-feedback protocol, source CH transmits to relay and the destination and then ceases its transmission after receiving ACKs of successful reception from relays hoping that destination can successfully receive the remaining information from relays. In both aforementioned protocols, all the nodes use their own fountain coding routine and different orthogonal channels to encode and transmit their data simultaneously. We study their performance in Rayleigh fading channel and show that by employing cooperative protocols the performance of data transmission can be dramatically improved than the direct communication between two neighboring CHs.

However, in conventional cooperative communications at a particular time a relay can only serve a single source. Hence employing cooperative communications alone may make the system bandwidth and power inefficient as well as increase end to end delay in multi-hop communications. Therefore we apply network coding at each relay. In particular, we employ network coding with fountain code and develop joint network-fountain coding (JNFC) schemes which can effectively combat the detrimental effect of wireless fading channel by seamlessly coupling fountain and network paradigms. The idea behind this is to coupling the network and channel coding techniques simultaneously in the physical layer so that the redundancy in the network code can be used to support the channel code for better error protection. We compare the performance of JNFC with separated network and fountain coding (SNFC) where the redundancy provided by network coding is only useful if channel coding is succeeded.

Finally, we focus on reliable uplink transmission between machine type devices and eNB where M2M devices transmit to eNB directly or with the help of the relay nodes. We investigate both type-I and type-II enhanced LTE-A network and propose a set of joint network and fountain coding schemes to enhance the link robustness. In particular, LTE uses systematic Raptor codes at the application layer for broadcast and multicast transmissions. However, LTE standardized Raptor codes can not provide effective symbol diversity for network coding due to its poor degree distribution. We develop several LT code based joint network channel fountain schemes for uplink transmissions between CHs and eNB. We evaluate the performance of the proposed schemes in terms bit error rate(BER) and transmission efficiency.

0.5 Contributions and Novelty of the Thesis

Guided by the objectives presented in Section 0.3 and using the methodology proposed in Section 0.4, this thesis makes the following important novel contributions:

- We identify the CH-to-CH channel as a bottleneck in clustered M2M network, and present a full solution to support fountain coded cooperative communications to facilitate the data transmission between two neighboring CHs. The proposed solution consists of several schemes from relay selection to cooperative strategy, while taking into account user's QoS requirements;
- We develop rateless coded incremental relay selection (RCIRS) algorithm that takes into account the link of the node between source CH and destination CH, and then picks up a minimum number of participating relays opportunistically while meet the QoS requirements;
- To accommodate different scenarios, we develop source-feedback and non-source-feedback cooperative communication protocols based on fountain coding. We evaluate the performance of cooperative protocols using the metric of transmission efficiency rather than outage probability. This strategy is more realistic since in rateless coded systems the probability of outage is always driven to zero as long as there is no constraint on decoding delay;

- Using cooperative communications alone may make the system bandwidth inefficient. To overcome the bandwidth bottleneck and improve the link robustness we integrated network coding with cooperative communications. In particular, we couple network and fountain coding techniques simultaneously in the physical layer so that the redundancy provided by network coding can be used to support the channel coding for better error protection;

- In legacy LTE network Turbo code is implemented as channel coding technique in the physical layer. Turbo code is a fixed rate that means the code rate is selected before the transmission. In poor channel condition if one or more retransmission are required to ensure a successful decoding, then 8 ms of delay is incurred for every retransmission that make the current LTE HARQ process unsuitable for many ultra reliable M2M communications. To solve this problem we develop a set of joint network fountain coding schemes for reliable uplink transmission between M2M devices and eNB and evaluate their performance in terms of BER and transmission efficiency.

0.6 Thesis Outline

The rest of the thesis is organized as follows. Chapter 1 provides the preliminaries on the required technical background for understanding the research area addressed in this thesis. The chapter also presents variety of techniques that have been found in literature on error recovery and mitigating the fading effect of wireless channel. Chapter 2-4 show the contribution of this work.

In Chapter 2, we develop a set of schemes to support the cooperative communication between two neighboring clusters. Our solution covers many aspects from relay selection to cooperative formation to meet the user's QoS requirements. We first review the existing work in Section 2.2 and introduce our system model of cooperative communications over clustered M2M networks in Section 2.3 . We assume that the network consists of two or more types of nodes with respect to battery supply and functionality. The network is then further partitioned into a number of clusters, where more powerful nodes have been chosen as CHs. CHs emulate the role of the base stations in their respective clusters, and inter cluster traffics are forwarded through the CHs to the LTE-A network. We then present the relay selection algorithm in Section 2.4. When a large number of non-CH nodes are available to be a relay, using exhaustive search to find out the optimum set requires huge computational effort. Considering the cost and performance issues in M2M network, we propose a relay selection algorithm namely rateless coded incremental relay selection (RCIRS) algorithm that fulfil the target data rate requirements between two clusters by employing minimum number of relay nodes. Our proposed relay selection algorithm is based on greedy technique that includes nodes in cooperative group one by one so that the cooperative data rate is incrementally increased until to meet the target data rate requirements. After that our proposed fountain coded cooperative communication protocols are presents in Section 2.5. We propose source-feedback and non-source-feedback protocols using Raptor code (Etesami & Shokrollahi, 2006) and compare the performance of theses protocols under different relay selection methods in Section 2.6.

In Chapter 3, to enhance the link robustness and error protection, we develop several joint network and fountain coding schemes that exploit the redundancy and network collaboration. In particular, we consider a cooperative system with two sources, two relays and one destination where the sources encode the message using LT code and broadcast to the destination and relays. While the relays first decode the information and then transmit to the destination after network and fountain coding. For information combining at relays we employ Random Linear Network Coding (RLNC) and Modified LT coding (MLT). We study their performance in different network scenarios. Moreover, we compare the performance of JNFC with separated network and fountain coding (SNFC) where the redundancy provided by network coding is only useful if channel coding is succeeded.

In Chapter 4, we focus on reliable uplink transmission between M2M devices and LTE eNB. In the beginning of this chapter we give a brief discussion about LTE uplink transmission, LTE channel coding and HARQ scheme and the limitation of the LTE-A technology to provide ultra reliable machine type communications. After that we present the state of the art to provide reliable M2M communications in LTE. We found that the current research efforts have mainly focused on efficient channel access, resource allocation and improving LTE-HARQ schemes. Other than previous works we focus on channel coding schemes and propose a set of joint network and fountain coding (JNFC) schemes for reliable uplink transmission between M2M devices and evolved NodeB (eNB). We believe due to the inherent characteristics of fountain code, it is very much advantageous for M2M communications. Like Turbo code it does not required sophisticated channel state information. In this chapter we consider both type I and type II enhanced LTE-A network and present numerical results to demonstrate the performance of the proposed JNFC schemes in terms of BER and transmission efficiency.

Finally, we conclude the thesis and discuss potential future work in Chapter 5.

0.7 List of Publications

The complete list of publications associated with this research work is presented below:

0.7.1 Journals:

Nessa, A., Kadoch, M.& Rong, B. (2016). Fountain coded cooperative communications for lte-a connected heterogeneous m2m network. *Ieee access*, 4, 5280-5292. doi: 10.1109/ACCESS.2016.2601031.

Nessa, A. & Kadoch, M. (2016). Joint network channel fountain schemes for machine type communications over lte-advanced. *Ieee internet of things journal*, 3(3), 418-427. doi: 10.1109/JIOT.2015.2497311.

Nessa, A., Kadoch, M. & Rong, B. (2014b). Efficient and reliable communication in wireless relay networks using joint network channel fountain. *JCM*, 9(8), 597–606. doi: 10.12720 /jcm.9.8.597-606.

0.7.2 Conferences:

Nessa, A., Kadoch, M., Hu, R. Q. & Rong, B. (2012, Dec). Towards reliable cooperative communications in clustered ad hoc networks. *Global communications conference (globecom)*, 2012 ieee, pp. 4090-4095. doi: 10.1109/GLOCOM.2012.6503757.

Nessa, A., Kadoch, M. & Rong, B. (2014a, Feb). Joint network channel fountain scheme for reliable communication in wireless networks. 2014 *international conference on computing, networking and communications (icnc)*, pp. 206-210. doi: 10.1109/ICCNC. 2014.6785332.

CHAPTER 1

BACKGROUND AND LITERATURE REVIEW

1.1 Introduction

This chapter provides a brief overview of M2M communications, LTE-A network and the ongoing development of LTE-A for M2M communications. It also presents the tools used in the literature for reliable data transmission in wireless communications and the related works.

1.2 Background

1.2.1 Machine to Machine Communication (MTC)

M2M communications is the key enabling factor of the practical realization of Internet-of-Things (IoTs). IoT is envisioned as a worldwide network of interconnected objects, where objects are equipped with tiny identifying devices, and intelligently transfers data over the network without any human intervention. This vision of IoT represents a future where billions of everyday objects of surrounding environments will be integrated with Internet to provide various services.

In the literature, several reports have appeared on the prediction of the market growth of the M2M devices in the next few years. For example, Cisco thinks approximately 50 billion of devices will be connected to the internet by 2020. Due to the emerging use of intelligent sensors and actuators in wide range of applications, it is obvious that M2M communications will initiates new market and create new stream of revenue for telecom operators, device manufacturers, M2M enablers and application developers. In order to grasp the full opportunities offered by global M2M market over cellular networks, 3GPP have initiated their working groups for facilitating such applications through various releases of their standards .

1.2.1.1 A General M2M Architecture

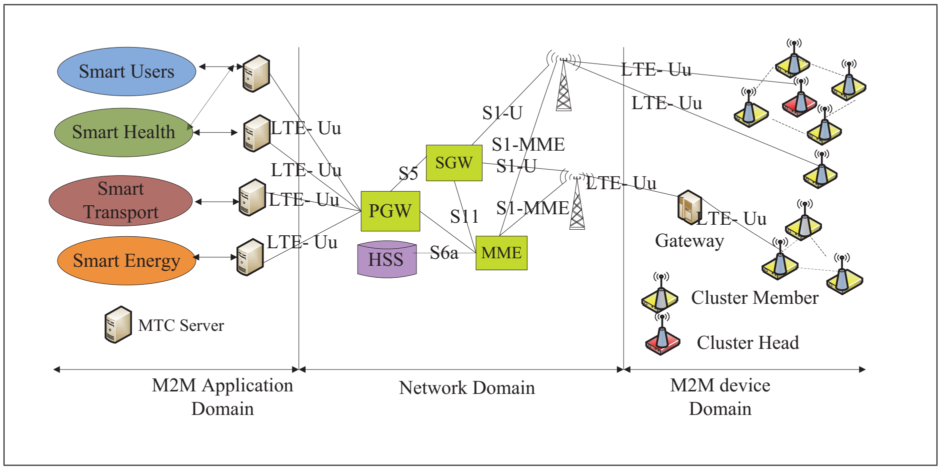


Figure 1.1 A general architecture of a M2M network

A generic M2M architecture consists of three distinct domains: M2M domain, network domain and application domain as shown in Figure 1.1.

- The M2M Domain: This domain consists of a large number of M2M devices those are connected with the M2M gateway in a single-hop or a multi-hop manner. An M2M device is equipped with multiple sensors to collect different types of sensed data and a wireless transceiver to transmit sensed data to the M2M gateways. M2M gateways act forward the sensed data to the communication network. M2M area networks also called subnets, make the communication path between the M2M devices and the M2M gateways. ZigBee, Meter bus (M-BUS), Bluetooth, Short-range device (SRD), Ultra wide band (UWB) are some of the major technologies used in subnet. In some scenarios, M2M devices also have the capability to directly access network domain. Like wireless sensor networks, M2M device domain can be classified into two broad types; homogeneous and heterogeneous M2M networks. In homogeneous networks all the M2M devices are identical in terms of battery energy and hardware complexity. These M2M devices are connected to the cellular network via gateway. On the other hand, in a heterogeneous M2M network, two or more different types of nodes with different battery energy and functionality are used. Devices are equipped with both a long range interface, through which they connect to, e.g., a cellular network or satellite system, and a short range interface, which is used to create an ad-hoc capillary network among the terminals;

- Network Domain: The major functionality of this domain is to create a communication path between the M2M devices and application servers. The communication network domain can be wired (Ethernet) or wireless cellular network (i.e., GSM, GPRS, EDGE, 3G,LTE-A, WiMAX, etc.). Although wired connection provide high reliability, high rate, short delay, and high security but due to its cost, maintenance, lack of scalability it may not be appropriate for the M2M communication applications. On the other hand, well established infrastructure, extensive coverage and mobility/roaming support offered by cellular network make it a potential candidate for providing the last mile M2M connectivity. Therefore, in this research work, our focus is on the M2M communications based on 3GPP LTE/LTE-A mobile networks. The LTE network includes Evolved Universal Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC). The EPC consists of a mobility management entity (MME), a serving gateway (SGW), and a packet data network gateway (PDNGW) together with a home subscriber server (HSS);

- Application Domain: In the MTC application domain, MTC users employ one or more MTC servers to control a large number of MTCDs in order to implement various MTC applications. An MTCD can communicate with an MTC server or other MTCDs using the 3GPP bearer services provided by the LTE network. When connected to the LTE network, the MTCD needs to be authenticated and authorized by the network before establishing any communication with the MTC server (Cao *et al.*, 2012).

According to the European Telecommunications Standards Institute (ETSI) M2M architecture and the network improvements for MTC developed by the 3GPP (Zheng *et al.*, 2014) three types of MTC access methods are defined, such as direct access, gateway access and coordinator access.

- Direct access: Like a user equipment device (UE), an MTC can directly access an evolved NodeB (eNB) in cellular networks without any intermediate device. However, this access method may lead to traffic overload in the presence of a large number of MTC devices even though small amount of data is transmitted by each MTC device;
- Gateway access: M2M gateways act like bridge between M2M devices and the communication network. An M2M gateway is equipped with more functional capabilities than the ordinary M2M devices. In gateway access intelligent sensors/ M2M devices sense the environment and sends their data to the MTC gateways. M2M gateways only collects the data from M2M device but do not generates its own data traffic and forward the aggregated data to the eNB;
- Coordinator access: In some cases, MTCDs can be grouped and one MTCD in each group plays the coordinator role to forward its group members data towards eNB. This type of access method reduce congestion on the air interface of the cellular network. It also reduces the power consumption of the overall M2M device domain. Therefore the life time of the battery-constrained devices is prolonged.

1.2.1.2 Features of M2M Communications

M2M applications are very much different from the conventional human-to-human (H2H) communications. Some of the key features and service requirements are highlighted below.

- Energy constrained: In most cases, M2M devices are battery powered device and very often deployed in remote areas where battery replacement is not easily possible. Therefore, energy efficiency is a critical issue for M2M communications to prolong network lifetime;
- Small payload: In most M2M applications M2M devices send or receive small amounts of data. Usually the ratio between payload and control information is low. Hence, the network must be able to support small amounts of data with minimal network impact (e.g., signalling overhead, resource utilization,etc.);

- Little or no mobility: M2M devices are generally static, move infrequently, or move only within a specific area;
- Lossy in nature: Reliable data delivery is no less important than energy efficiency in M2M communications because M2M devices might be deployed in challenging wireless environments where links are prone to frequent failures;
- Device heterogeneity: There are already a large number of M2M applications and use cases due to which a large variety of devices with diverse service requirements have emerged. This creates a major challenge for inter operability as well as the generalization of M2M;
- Delay sensitive: M2M applications that require extremely fast access to the network and cannot tolerate much delay (e.g., smart grid recovery operations, and emergency automatic shutdown of a gas pipeline in case of earthquake or other calamity);
- Security vulnerabilities: Security is an important issue for successful applications of M2M communications. Since a large number of M2M devices are dispersed in diverse environment, they are fundamentally vulnerable to both physical and cyber attacks. In (3GPP-TR-33.812, 2010), the categories of possible attacks in M2M communication are specified as physical attacks, compromise of credentials, denial-of-service attacks, user data and identity privacy attacks. Since, M2M devices are running autonomously, it is quite difficult to detect the attack quickly. Moreover, due to the resource limitation of M2M devices in terms of energy and processing capability, existing security mechanisms may not be feasibly used for M2M communications. Hence, the role of security becomes critical in M2M communications.

1.2.2 LTE Overview

1.2.2.1 Evolution of LTE to LTE-A and Beyond 4G Systems

The 3rd Generation Partnership Project (3GPP) was established in 1998 to create a collaboration among different telecommunications associations. In release 8, 3GPP specified evolved packet system (EPS) architecture that contain two major work items, namely system architecture evolution (SAE) and long term evolution (LTE), that led to the specifications of the evolved packet core (EPC), evolved universal terrestrial radio access network (E-UTRAN) and evolved universal terrestrial radio access (E-UTRA), each of which corresponds to the core network, radio access network, and air interface of the whole system, respectively.

LTE provides extensive support for spectrum flexibility, supports both frequency division duplex (FDD) or time division duplex (TDD) mode, also referred to as LTE FDD and TD-LTE, and targets a smooth evolution from earlier 3GPP technologies such as TD-SCDMA and WCDMA/HSPA as well as 3GPP2 technologies such as CDMA2000. According to the specification defined in Rel-8, LTE can provide downlink and uplink peak rates up to 300 Mbit/s and 75 Mbit/s, respectively, a one-way radio-network delay of less than 5 ms, and a significant increase in spectrum efficiency. The main key technology aspects of LTE are:

- Orthogonal Frequency Division Multiple Access (OFDMA) based multiple access schemes for both LTE FDD and TD-LTE Scalable bandwidth up to 20 MHz;
- Support for Multiple Input Multiple Output (MIMO) antenna technology;
- New data and control channels;
- New network and protocol architecture (two node, IP based).

In July 2008, international telecommunication union (ITU) published the formal specification of the fourth generation wireless, known as the international mobile telecommunications advanced (IMT Advanced). In order to be a true 4th generation (4G) technology, LTE was enhanced to meet the IMT-Advanced requirements issued by the ITU. The necessary improvements are specified in 3GPP Release 10 and also known as LTE-Advanced. The relationship among the requirements of LTE, LTE-Advanced, and IMT Advanced are shown in Table 1.1. LTE-A builds on the LTE OFDM/MIMO architecture to further increase data rate. LTE-Advanced increases peak data rates towards 1 Gbit/s in the downlink and 500 Mbit/s in

Table 1.1 LTE, LTE-Advanced, and IMT-Advanced performance targets for downlink (DL) and uplink (UL)
Taken from Akyildiz et al.(2014)

Item	Transmission path	Antenna configuration	LTE (Rel. 8)	LTE-Advanced	IMT-Advanced
Peak data rate	DL	8×8	300 Mbps	1Gbps	1Gbps
	UL	4×4	75 Mbps	500 Mbps	-
Peak spectrum efficiency (bps/Hz)	DL	8×8	15	30	15
	UL	4×4	3.75	15	6.75
Capacity (bps/Hz/cell)	DL	2×2	1.69	2.4	–
		4×2	1.87	2.6	2.2
		4×4	2.67	3.7	–
	UL	1×2	0.74	1.2	–
		2×4	–	2.0	1.4
Cell-edge user throughput (bps/Hz/cell/user)	DL	2×2	0.05	0.07	–
		4×2	0.06	0.09	0.06
		4×4	0.08	0.12	–
	UL	1×2	0.024	0.04	–
		2×4	–	0.07	0.03

Table 1.2 Comparison between 5G targets and LTE-Advanced Pro

Item	**5G Targets**	**LTE Advanced Pro**
3GPP Standard Release	3GPP Release 15 and Beyond	3GPP Release 13 and beyond
Total carrier Bandwidth	100MHz carrier BW for gigabit backhaul and 500MHz for multi-gigabit backhaul	640 MHz, aggregates up to 32 carriers each of 20MHz bandwidth
Data rate	About 10 Gbps	More than 3 Gbps
Operating frequency	6GH	Below 1 GHz up to close to 100 GHz
Latency	Less than 1ms round trip time	Less than 2ms round trip time and less than 1ms one way delay

the uplink. In order to be a true 4th generation (4G) technology, in LTE-A the following technology components are adopted:

- Carrier aggregation;
- MIMO extension (up to DL: 8x8; up to UL: 4x4);
- Uplink access enhancements (clustered SC-FDMA and simultaneous data and control information (PUSCH and PUCCH) transmission;
- Improving cell edge performance (enhanced inter-cell interference coordination (eICIC), relaying).

After the introduction of the 4G LTE-Advanced standard in 3GPP Releae 10, LTE-Advanced has continued to evolve through several releases. Release 11 was started and further enhancements were included to the basic LTE-Advanced technologies developed for Release10. The major contribution during Release 11 were cooperative multipoint transmission and reception (CoMP), enhanced inter-cell interference cancellation (eICIC) and mobility management enhancements. After the completion of Release 11 in early 2013, the standardization work began on 3GPP Release-12. The primary goal of Release 12 is to provide mobile operators with new options for increasing capacity, extending battery life, reducing energy consumption at the network level, maximizing cost efficiency, supporting diverse applications and traffic types, enhancing backhaul and providing customers with a richer, faster and more reliable experience.

The research community is now increasingly looking beyond 4G and working towards future 5G technologies. ITU-R has recently finalized work on the "Vision" for 5G systems, which includes support for explosive growth of data traffic, support for massive numbers of machine type communication (MTC) devices, and support for mission critical and ultra-reliable and low latency communications (ITU-R-M.2083, 2015). The overall 5G wireless-access solution consisting of LTE evolution and new radio access technology. LTE evolution will focus on backwards-compatible enhancements in existing spectrum up to 6 GHz, while new radio access technology will focus on new spectrum where LTE is not deployed. On the path to 5G, the Release 13 and Release 14 known as LTE Advanced Pro standard is already using carrier aggregation, massive MIMO and simultaneous use of licensed and unlicensed spectrum techniques to provide Gigabit LTE. To provide high data rate Release 13 includes three major technology categories such as full dimension MIMO (FD-MIMO) that aims to drastically increase spectral efficiency via the use of a large number of antennas at the base station(Massive MIMO), utilizing unlicensed spectrum while guaranteeing coexistence with existing devices and enhanced carrier aggregation (eCA). Another important focus of Release 13 was cost reductions for MTC devices that can also support extended coverage. The focus of release 14 includes the extension of current massive MIMO framework to an even larger number of antennas, reducing the transmission time interval up to .5ms or less and supporting massive MTC. Table 1.2 presents the relationship among the requirements of LTE-Advanced Pro and 5G.

1.2.2.2 3GPP Network Architecture: E-UTRAN Overview

LTE-A network comprises two parts, i.e., the EPC or core network (CN) and the Radio access network(RAN). The EPC is responsible for overall control of mobile devices and establishment of Internet Protocol (IP) packet flows. RAN consists of base stations (BSs) that are referred to as evolved node base stations (eNBs) (3GPP-TR-36.300, 2012) that is responsible for wireless communications and radio access, and provides an air interface with user plane and control plane protocol terminations toward the user equipment (UE) and M2M devices. In this subsection we provide an overview of the E-UTRAN architecture.

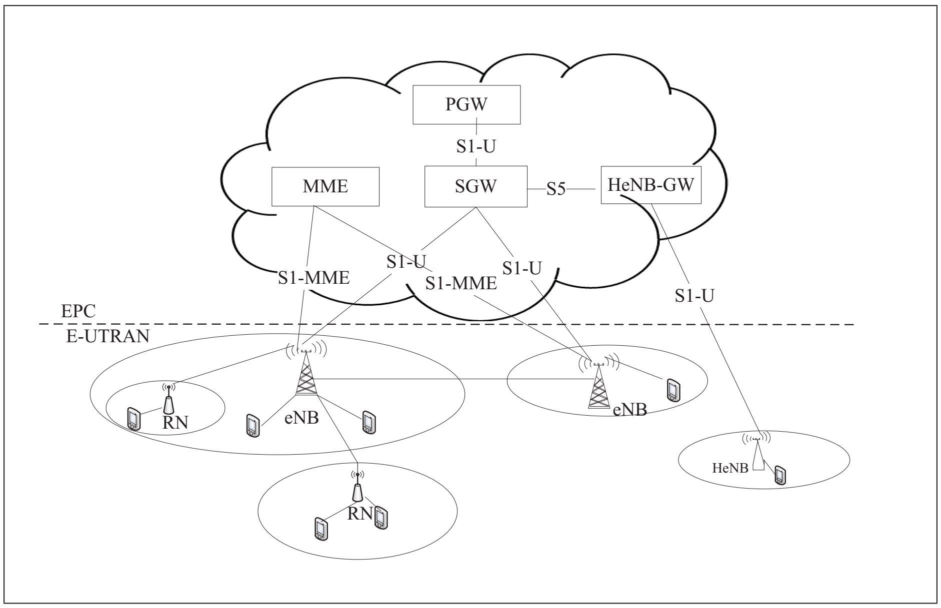


Figure 1.2 LTE-Advanced E-UTRAN overview

The architecture of the E-UTRAN for LTE-A is shown in Figure 1.2. The E-UTRAN handles communication between the UEs and the EPC. The main part in the E-UTRAN architecture is enhanced Node B (eNodeB or eNB). The eNB provides the air interface with user plane and control plane protocol terminations towards the UE. Each of the eNBs is a logical component that serves UEs of one or several E-UTRAN cells. The serving eNB (SeNB) is the eNB which currently provides services to a UE. Moreover, the eNB controls low-level operations such as handover functions. The eNB is connected to the EPC through the S1 interface. The connection among the eNBs is accomplished through the X2 interface which is mainly used for exchanging information such as packet forwarding during handover. Moreover, eNB establishes the communication path between UE and mobility management entity (MME). On the air interface (Uu), the UE high-level functionalities are controlled by the MME . The Uu interface is divided into two levels of protocols. One is called access stratum (AS) whereas the second is the nonaccess stratum (NAS). The MME high-level signaling lies in the NAS level but is transported within the network using AS protocols.

As mentioned earlier, the eNB provides the E-UTRAN with the user and control plane termination protocols. User plane protocols support routing of users data between UEs and S-GWs whereas control plane protocols are used for exchanging signaling messages between various devices within the network. Figure 1.3 gives a graphical overview of both protocol stacks. In the user plane, the protocols include packet data convergence protocol (PDCP), radio link control (RLC), medium access control (MAC), and physical layer (PHY) protocols. The protocols of the control plane include the radio resource control (RRC)protocols.

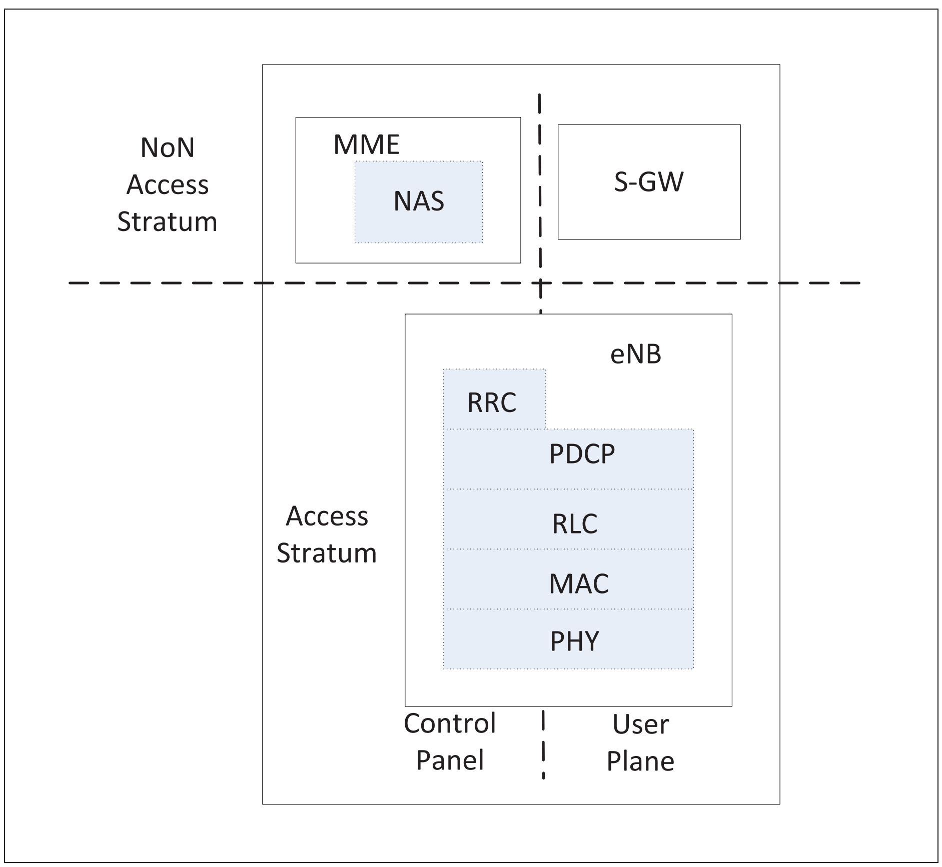


Figure 1.3 User and control panel protocol stack

A high level overview of the protocol stack is depicted in Figure 1.3. The major functionalities of these protocols are given below (3GPP.36.323, 2009; 3GPP.36.322, 2010; 3GPP.36.321, 2010; 3GPP.36.331, 2010).

- Access stratum (AS): UEs are capable to access the capabilities and services of communication networks through the AS protocols. The functionalities of the radio AS protocols includes dynamic allocation of radio resources to UEs(UL/DL), controlling bearers, traffic management, radio admission control, and handover management;
- Non-access stratum (NAS): The NAS functionalities include the establishment of radio connections between the network and the UE. In addition, registration, authentication, local registration management,bearer context activation/deactivation, and mobility management are also included in the functionalities of the NAS protocols;
- Radio resource control (RRC): The RRC protocol layer handles the control plane signaling between the UE/M2M and eNB. The main services and functions of the RRC sublayer include broadcast of system information related to the NAS and AS. In addition, establishing/releasing RRC connections, Initial security activation , mobility functions, paging, QoS management, and direct messaging between UE and NAS;
- Packet data convergence protocol (PDCP): This layer performs IP header compression and decompression, and transfers user plane or RRC data. Maintenance of PDCP sequence numbers for radio bearers and in-sequence delivery of upper layer packet data units (PDUs) are other functions of PDCP layer. In addition, duplicate detection of lower layer session data units (SDUs), ciphering and deciphering of user plane data and control plane data, and integrity protection of control plane data are performed in this layer;
- Radio link control (RLC): The RLC protocol layer exists in UE/M2M and eNB. The functionalities of this layer include segmentation of data packets according to the available size of the transport block. Moreover this layer performs error correction through automatic repeat request (ARQ). In addition, re-segmentation and reordering of RLC data PDUs, RLC

re-establishment, and error detection and recovery are the other functions of this protocol layer;

- Medium access control (MAC): The MAC layer functionalities include mapping between logical and control channels, RLC SDUs multiplexing/demultiplexing, scheduling, error corrections using hybrid ARQ (HARQ), priority handling of local channels, segmentation and reassembly of upper layer PDUs, padding, and ciphering;

- Physical layer (PHY): Physical Layer carries all information from the MAC transport channels over the air interface. The PHY layer provides data transport services to the upper layers such as synchronization of time and frequency, physical channel modulation/demodulation, encoding/decoding of transport channels, MIMO operations, and transmit diversity.

  Moreover, E-UTRAN also provides home eNB (HeNB) services. HeNB is an eNB which is particularly used for indoor coverage improvement. It can be connected either directly to the EPC in the same way as a standard eNB or via a gateway that caters for additional support for a large number of HeNBs. In addition, the 3GPP LTE-A encompasses relay nodes and sophisticated relaying strategies for network performance augmentation. The aim of this new technology is to offer large coverage, high data rate, and better QoS performance and fairness for different users.

1.2.2.3 3GPP Network Architecture: EPC Overview

The EPC is a "flat" all IP-based core network that can be accessed through 3GPP radio access (e.g.,WCDMA, HSPA, and LTE/LTE-A) and non-3GPP radio access (e.g., WiMAX and WLAN) and allows handover procedures within and between both access types. This system is considered as “flat” because from a user-plane perspective there are only the eNBs and the gateways that reduced complexity. The main components of EPC are presented in Figure 1.2 and a brief discussion on the main components are given bellow:

- Mobility management entity (MME): The MME is a key control plane element for the LTE/LTE-A access network. The main functionalities of MME include managing security functions (authentication, authorization, and NAS signaling), roaming, handover, and handling idle mode user equipment. It is also in charge of choosing the Serving Gateway (S-GW) and packet data network gateway (P-GW) for an UE/M2M device at an initial attach. The S1-MME interface connects the EPC with the eNBs;

- Serving gateway (S-GW): The S-GW is the termination point of the packet data network interface towards E-UTRAN. S-GW maintains data paths between eNodeBs and the PDN Gateway (P-GW). It is connected to the eNB through S1-U interface and to the P-GW through S5 interface. Each UE/M2M device is associated to a unique S-GW, which will be hosting several functions. When terminals move across areas served by eNodeB elements in E-UTRAN, the SGW serves as a local mobility anchor. This means that packets are routed through this point for intra E-UTRAN mobility and mobility with other 3GPP technologies, such as 2G/GSM and 3G/UMTS;

- Packet data network gateway (P-GW): The P-GW provides connectivity from the UE/M2M device to an Packet Data Network (PDN) by assigning an IP address from the PDN to the UE/M2M device. Moreover, P-GW provides security connection between UEs/M2M devices by using Internet protocol security (IPSec) tunnels between UEs/M2M devices connected from an untrusted non-3GPP access network with the EPC.

1.2.2.4 LTE-Advanced Relaying

With orthogonal frequency-division multiplexing (OFDM), multiple-input multiple-output (MIMO), capacity-approaching codes, a pure packet-switched core network, and other technologies, LTE aims to meet the high data rate and spectral efficiency demand for future applications. However the use of these technologies such as MIMO, OFDM and advanced error correction techniques improve throughput under many conditions, but not fully mitigate the problems experienced at the cell edge. Therefor user closer to the eNB receives better performance in terms of throughput and power savings than the cell edge users.

As a solution to overtake the above mentioned problem in LTE Release 10, 3GPP has begun supporting relay technology for relaying radio transmissions between eNB and UE. LTE-A relays are low power nodes that can be deployed underlying macro eNBs to address the need for coverage and capacity improvements at cell edge or to extend the coverage in various types of locations where fixed link backhaul are difficult to implement. Two types of relays (RNs), i.e., type I and type II also know as " non-transparent" and "transparent" relay, respectively are mainly discussed for LTE-advanced systems in the 3rd Generation Partnership Project (3GPP) (3GPP.36.814, 2010). This classification is mainly based on the relay's ability to generate its own cell control message (Loa *et al.*, 2010).

- Type I relay has its own cell and physical cell identification (ID), transmits synchronization signals and performs resource allocation for the UEs and appears as a regular eNB to all the UEs of its cell. The typical usage scenarios of type I relay include coverage extension, e.g., hotspot and deadspot, and group mobility, e.g., on a train or bus;
- Type II relays do not have their own physical identity and as the part of eNB they are deployed underlying macro eNBs to facilitate the uplink and downlink transmission between cell edge UEs and eNB. The type II relay exploits the cooperative mode of relay and participates in data forwarding between UE and eNB while the UE is unaware of their existence.

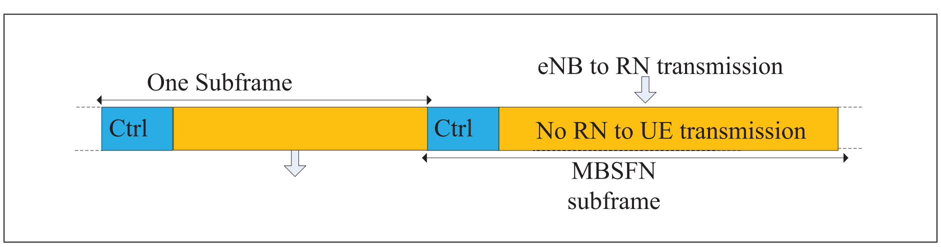


Figure 1.4 RN-to-UE transmission using normal subframes (left) and eNB-to-RN transmission using configured MBSFN subframes (right)

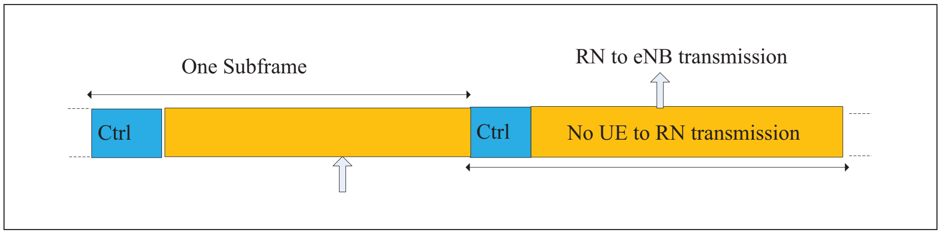


Figure 1.5 UE-to-RN transmission (left) and RN-to-eNB transmission (right) through uplink grant scheduling

However, RN does not have any wired connection like eNB to the EPC or the core network. The associated eNB of the RN donates its radio resources to the RN's backhaul link and is known as Donor eNB (DeNB). The donor-relay link may operate on the same frequency as the relay-terminal link (inband relaying) or on a different frequency (outband relaying).

To ensure the relay is not transmitting on the access link while it is receiving on the backhaul link, some subframe should be reserved for the backhaul link. In Release 10 a gap in the relay-to-terminal transmissions to allow for reception of donor-to-relay transmissions is created using multicast-broadcast single-frequency network (MBSFN) subframes as shown in Figure 1.4. In an MBSFN subframe the first one or two OFDM symbols in a subframe are transmitted as usual carrying cell-specific reference signals and downlink control signaling and the remaining part is used for the donor-to relay communication. MBSFN subframes were originally intended for broadcast support in release 8 but has later been used for relaying support. Like the transmission gap, reception gaps is required in the access link in order to transmit on the backhaul from relay to eNB. Such gap is created by blocking UE transmission through the proper scheduling of uplink transmission as shown in Figure 1.5.

1.2.3 Machine Type Communications Over 3GPP LTE/LTE-A Networks

1.2.3.1 3GPP MTC Reference Model

Over 3GPP LTE/LTE-A Networks, the end-to-end communications, between MTC devices and application server uses 3GPP provided services and the optionally services provided by external services capability server (SCS). The 3GPP system provides several services to support MTC communications including transport, subscriber management as well as various architectural enhancements. 3GPP envisages three types of communication model between MTC device and the MTC server that are given bellow:

- Direct model: It is the most straight forward model where the application server (AS) connects directly to network and perform data communication with UE without using any SCS service as shown in Figure 1.6;
- Indirect model: In this model AS connects indirectly to the network through the services of a SCS in order to utilize additional value added services for MTC (e.g. control plane device triggering). The SCS can be located either outside of the operator domain (MTC service provider controlled communication), or inside the operator domain (3GPP operator controlled communication) and consider as an internal network function;
- Hybrid model: In this model AS uses the direct model and indirect models simultaneously. AS uses the direct model to connect directly to the operator's network to perform direct user plane communications with the UE. However it also uses SCS services. From the 3GPP network perspective, the direct user plane communication from the AS and any value added control plane related communications from the SCS are independent and have no correlation to each other even though they may be servicing the same MTC Application hosted by the AS.

1.2.3.2 3GPP LTE/LTE-A Architectural Reference Model for M2M Communications

3GPP LTE/LTE-A architectural reference model for M2M device as specified by the 3GPP in their Release 11 specifications is illustrated in Figure 1.7. It is shown that M2M devices connect with 3GPP network via Um/Uu/LTE-Uu interfaces. The description of the MTC related reference point are given bellow:

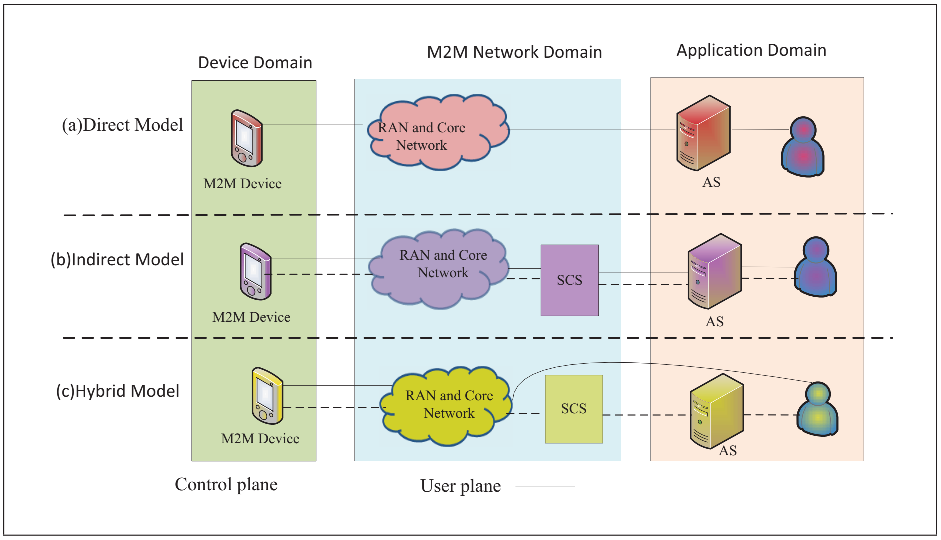


Figure 1.6 3GPP direct, indirect, and hybrid models for MTC communications

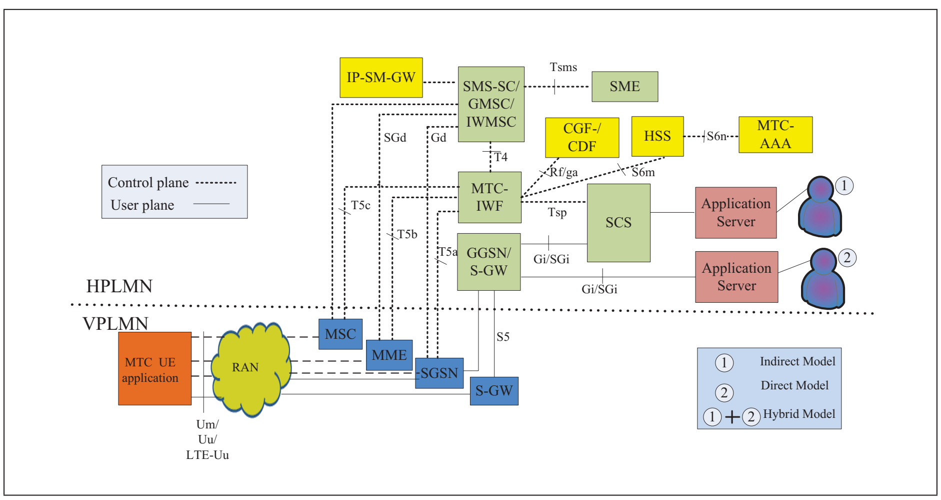


Figure 1.7 3GPP LTE/LTE-A architecture reference model for M2M communications
Adapted from Ghavimi & Chen (2015)

- Tsms: It enables the communication between an entity lying outside the 3GPP network and MTC using SMS. Tsms is a non-standardized interface that encompasses various proprietary short message service-service center (SMS-SC) to shortmessage entity (SME) interfaces [29]. Tsms can be used to send a trigger to an M2M device encapsulated in a mobile terminated-SMS (MT-SMS) as an over-the-top application by any network entity (e.g., SCS) acting as an SME;

- Tsp: It connects MTC-IWF to SCSs to provide several functionalities such as device triggering;
- T4: It is used by the MTC interworking function (MTC-IWF) to route a device trigger as an MT-SMS to the SMS-SC in the HPLMN;
- T5a It enables communication between MTC-IWF and serving SGSN;
- T5b: It establishes the communication path between MTC-IWF and serving MME;
- T5c: It creates the communication link between MTC-IWF and serving MSC;
- S6m: Reference point used by MTC-IWF to interrogate HSS/HLR. The MTC-IWF uses S6m interface to interrogate the home subscriber server (HSS)/home location register (HLR) for mapping an external identifier or a mobile station integrated services digital network (MSISDN) to the international mobile subscriber identity (IMSI), authorizing a device trigger to a particular M2M device, and retrieving serving node information;
- S6n: Reference point used by MTC-AAA to interrogate HSS/HLR. The MTC accounting, authorization, and authentication (MTC-AAA) server uses an S6n interface to interrogate HSS/HLR for mapping IMSI to external identifier(s) and vice versa.

The following 3GPP network elements provide the functionalities to support the indirect and hybrid models of MTC. As 3GPP notes, as further development of the MTC architecture takes place, further network elements may be defined in the future (Ghavimi & Chen, 2015)

- MTC-IWF: To support indirect and hybrid communication between of MTC devices and MTC server, one or more instances of an MTC-IWF reside in HPLMN. A MTC-IWF may be a stand alone entity or a functional entity of another network element. The MTC-IWF hides the internal PLMN topology and relays or translates signaling protocols used over

Tsp to invoke specific functionality in the PLMN. Another main task of MTC-IWF is to authorize the SCS before communication establishment with the 3GPP network. For the roaming scenario, the MTC-IWF shall have the connection with HSS and SMS-SC within the home network only and with serving SGSN/MME/MSC in the visited network;

- MTC- AAA: The MTC AAA uses an S6n interface to interrogate HSS/HLR for mapping the international mobile subscriber identity (IMSI) to external identifier(s) and vice versa. To support translation of the IMSI to External Identifier(s) at the network egress, an AAA function (MTC AAA) is used in the HPLMN. The MTC AAA may be deployed to return the external identifier(s) based on IMSI. Alternatively the MTC AAA may be deployed as a RADIUS/Diameter proxy between the GGSN/PGW and the AAA server in the external PDN;

- HSS/HLR: The Home subscriber server (HSS)/home location register (HLR) determines whether an SCS is allowed to send a trigger to a particular MTC. The triggering process terminates the S6m reference point where MTC-IWFs connect to the HLR/HSS and stores routing information (i.e. serving MME/SGSN/MSC address);

- SGSN/MME: The serving GPRS support node (SGSN) and mobility management entity (MME) receive device trigger from MTC-IWF and optionally stores it. SGSN encapsulate device trigger information in non-access stratum (NAS) message sent to the UE/M2M device, receives device trigger delivery success/failure status to MTC-IWF. Furthermore, SGSN performs security functions, access control and location tracking. It plays the role of MME and serving gateway (S-GW) in the EPC;

- MSC: The mobile switching center (MSC) server controls circuit-mode services. The MSC is mostly associated with communications switching functions, such as call set-up, release, and routing. It also performs a host of other duties, including routing SMS messages, conference calls, fax, and service billing as well as interfacing with other networks, such as the public switched telephone network (PSTN);

- GGSN/P-GW: The gateway GPRS support node (GGSN) or packet data network gateway (P-GW) may support the following functionality. Based on access point name (APN) configuration and unavailability of MSISDN and external identifier(s) in the GGSN/P-GW either queries an MTC accounting, authorization, and authentication (AAA) server for retrieval of external identifier(s) based on IMSI or routes RADIUS/Diameter requests for AAA servers in external packet data network (PDN). The GGSN function is similar to the P-GW in the EPC.

1.2.3.3 LTE-A Enhancement for Machine Type Communication (MTC)

MTC posses a different set of challenges on LTE cellular systems which is primarily built for mobile broadband communication. In order to fully support MTC, these challenges must be effectively addressed for the future broadband wireless communications. In this vein, 3GPP has already specified MTC features and the service requirements for MTC applications, and identified issues and challenges related to them (Shariatmadari *et al.*, 2015; 3GPP.23.368, 2011). Several improvements have been initiated in 3GPP targeting to enhance the LTE radio access network for M2M communications (3GPP.23.887, 2013; 3GPP.36.888, 2013; 3GPP.37.868, 2013; 3GPP.37.869, 2013).

In fact, 3GPP has started to integrate machine-type communication (MTC) into LTE-A from Release-10. The main focus of Release 10 and 11 was to protect the mobile network against overload as a result of the many IoT devices. The first set of enhancements specified in Release 10 includes MTC subscription and the overload and congestion control mechanism to handle the large amount of MTC devices (3GPP.23.401, 2011; 3GPP.24.301, 2011). The subscription control enables the MTC subscribers to activate/deactivate one or more MTC features in the subscription based on the operator policy. The overload and congestion control targets the prevention of being down the core network due to simultaneous transmission from many MTC devices. If an overload situation occurs, the MTC device receives an extended waiting timer and has to wait until its expiry before sending new request. Release-11 continued the support of MTC further by addressing device triggering and identification mechanisms.In addition, to provide support for the communication between the MTC application in the MTC UE connected to 3GPP network and the MTC application in the external network, the 3GPP architecture was enhanced in Release-11 (3GPP.23.682, 2012). That has been illustrated in Figure 1.7 and described in the previous section. Device triggering is required when an IP address for the device is not available or reachable by the service capability server (SCS)/application server (AS). In this case SCS send a device trigger request over the 3GPP network to trigger the device to perform application specific actions. The Release 11 specifications introduce a new identity known as the external identifier to identify the large number of MTC devices.

Since MTC devices are expected to be deployed in a large volume in remote areas with poor LTE coverage, reducing the cost and power as well as improve the coverage for such deployment are some of the the requirements to support M2M in LTE. A new category of low cost and low throughput devices is needed for many IoT applications. To improve the performance of MTC in LTE-A further, two features were specified in Release12: one is small data enhancement and another is power saving for ultra-long battery life. Release-12 introduced Low-cost M2M devices called Category-0 with reduced performance requirement that meets the needs of many machines. The main characteristics of Category-0 UE (3GPP.36.306, 2015) includes single receive (Rx) antenna, peak data rate 1 Mbps in downlink and uplink and half duplex mode. In Rel-12, a power saving mode namely discontinuous reception(DRX) was introduced. In this mode, the UE remains registered to but not reachable by the network for mobile terminating traffic. The UE can be viewed to be in the power-off or sleep mode, and will wake up only when there is data to send or after timer expiration and then goes to power-off or sleep mode while it remains registered. This feature is well suited for M2M services with timed transmissions such as remote sensing and smart grid.

3GPP Release-13 encompasses enhanced machine type communications (eMTC) and narrow band IoT (NB-IOT) to support a wide variety of IoT use cases. Many features of category 0 UEs such as a single receiver antenna, reduced soft buffer size, reduced peak data rate (1 Mb/s),and half duplex operation with relaxed switching time are maintained for eMTC. Some new features are also introduced to reduce the cost of the of the device. The new features are reduced transmission mode(only mode 1,2,6 and are supported), reduced number of blind decodings for control channel, allowing narrow band operation, no simultaneous reception. The maximum channel bandwidth of this narrow band is 1.08 MHz or 6 LTE resource block(RB). An eMTC always contained in this narrow bandwidth for the transmission and the reception of the physical channels.

NB-IoT further decrease the bandwidth requirements compared to eMTC. NB-IoT has a bandwidth of just 180 kHz. The smaller device bandwidth reduces peak data rate (around 50 kb/s for uplink and 30 kb/s for downlink) and increases latency. The design targets of NB-IoT include low-cost devices, high coverage (20-dB improvement over GPRS), long device battery life (more than 10 years), and accommodating massive number of devices (greater than 52K devices per channel per cell). Furthermore NB-IoT UEs only support limited mobility. The low data rate and reduce reduced mobility feature of NB-IoT targets the typical use cases of machine type communications(e.g. smart metering). NB-IoT can be deployed by using unused LTE resource blocks, free spectrum between neighboring LTE carriers (guard band) or stand-alone, for example in unused GSM carriers. Extended discontinuous reception (eDRX) is another improvement included in Release 13 for power saving of the devices. In Release 12 the maximum DRX time is 2.56 seconds. That targets the device that expects data only every 15 minutes and has relaxed delay requirements. In eDRX the timer can be configured in the range from some few seconds to some hours that can efficiently support M2M communications with low duty cycle. Release 13 introduced eDRX for both idle and connected mode. For idle mode, the maximum possible DRX cycle length is extended to 43.69 min, while for connected mode the maximum DRX cycle is extended up to 10.24 s (Rico-Alvarino *et al.*, 2016).

LTE is continuously evolving and introducing enhancement and new features in order to meet 5G requirements. In light of the advancement, Release 13 and onwards are known as LTE-Advanced Pro. Latency reduction, enhancement for MTC , operation in unlicensed spectrum and support for intelligent transportation systems are some of the major features that will be included in Release 14.

1.3 Literature Review

Reliable data transmission is a compelling demand of many M2M applications. M2M network consists of a large number of low-cost, low-power and multi-functional sensor nodes that are deployed in challenging wireless environment where the link is very often suffered by various dynamic environmental factors such as heat and humidity, obstacles and RF interference and results in high bit error rate and time-varying packet losses. In the literature a variety of tools have been found on error recovery and mitigating the fading effect of wireless channel. In this section we will review/discuss some techniques and corresponding research works on those topics.

1.3.1 Automatic Repeat Request(ARQ)

The automatic repeat request (ARQ) is the easiest and most frequently used technique to provide reliable transmissions. ARQ is a feedback-based mechanism where the receiver first detects the lost packets and then request the sender to retransmit those packets. Based on the retransmission strategies, there are three basic types of ARQ schemes: stop-and-wait ARQ, go-back N ARQ, and selective-repeat ARQ (Darnell, 1985). The most simplest ARQ scheme is stop-and-wait scheme. In a stop-and-wait ARQ error-control system, the transmitter sends a codeword to the receiver and waits for an acknowledgment. If the receiver detects an error in the transmitted packet then it send a negative acknowledgment (NAK) to transmitter. After receiving an NACK, the transmitter resends the packet and again waits for an acknowledgment. In this way retransmissions continue until the transmitter receives an ACK. Though this scheme is simple but inherently inefficient because of the idle time spent waiting for an acknowledgment of each transmitted codeword. The idle time spent during waiting can be reduced by allowing the transmitter to continue sending enough frame so that the channel is kept busy while the transmitter waits for ACK.

In go-back N ARQ, the transmitter sends a number of packets specified by a window size even without receiving an acknowledgement (ACK) from the receiver. The receiver process keeps track of the sequence number of the next packets it expects to receive and send ACK for every received packet. The receiver discards any packet that does not have the exact sequence number it expects. Once the transmitter has sent all the packets of its window, it will go back to the sequence number of the last ACK it received from the receiver process and fill its window starting from the last lost frame and continue the process over again. Though go-back N ARQ reduces the idle time but increases the frequency of retransmissions for each packet. Moreover, in channels that have high error rates the go-back N ARQ protocol is inefficient because it demands the retransmission of the packet in error and the subsequent packets. In order to make the ARQ scheme more efficient selective repeat ARQ scheme where the receiver informs transmitter about the missing packts by ACKs. The transmitter then individually retransmits the packts that have timed out.

Recently cooperative ARQ protocols (Tacca *et al.*, 2007; Lee *et al.*, 2010) has attracted much attention for sensor nodes that recharge their batteries by harvesting energy from the surrounding environment. In cooperative ARQ protocols, one or more cooperative nodes - instead of the source retransmit the lost packet to the destination when the earlier transmission attempt made by the source is unsuccessful. With this retransmission mechanism, it is as if sensor nodes could borrow energy from one another and balance their energy consumption to match their own battery recharge rate. In literature, a comprehensive studies has been found on the the performance of ARQ based protocols (Kim & Krunz, 2000; Badia *et al.*, 2006; Le *et al.*, 2007). (Kim & Krunz, 2000) analyzed the mean delay of a Markovian source using selective-repeat ARQ scheme. In L.Badia2006, the authors investigated the packet delay statistics of the selective-repeat ARQ in Markov channels. A queueing analysis of ARQ protocols together with adaptive modulation and coding strategies was presented in (Le *et al.*, 2007) using matrix geometric methods. In (Qiao *et al.*, 2009), energy efficiency of fixed-rate transmissions was analyzed under statistical queueing constraints when a simple ARQ scheme was employed in outage events.

1.3.2 Forward Error Correction (FEC) Codes

One of the conventional method to provide reliable communication is using redundant information to correct errors in the original information, that is also known as forward error correction (FEC). FEC can be used in the situations where retransmission relatively costly, impossible or too slow, such as in deep space or satellite communication (Xie *et al.*, 2010). It is also a desirable solution for reliable multicast application. The redundant information can be added either inside each packet, known as error-correction coding or across multiple packets, known as erasure-correction coding.

1.3.2.1 Channel Coding

Channel coding is a widely used error-correction technique in wireless communications. It is implemented at the physical layer by adding redundant information (e.g parity-check bits/symbols) inside each packet to recover erroneous information. The error recovery capability depends on the specific coding strategy and the amount of redundant bits/symbols. The major categories of FEC codes are convolutional codes and block codes. Convolutional codes are processed on a bit-by-bit basis. They are particularly suitable for implementation in hardware, and the Viterbi decoder allows optimal decoding. Block codes are processed on a block-by-block basis. Reed-Solomon, Turbo (Berrou *et al.*, 1993) and low-density parity-check (LDPC) codes are some most important block codes. Berrou et al. demonstrated the excellent performance of Turbo for data transmission over over additive white Gaussian noise (AWGN) channels (Berrou *et al.*, 1993; Berrou & Glavieux, 1996). Later the performance of Tubo code over Rayleigh fading channels was presented by (Hall & Wilson, 1998). Today's communication technologies utilize different coding schemes depending on their applications. The global system for mobile (GSM) communication employs convolutional codes with Viterbi algorithm at the decoder. The extensions of the GSM such as the GPRS1, EDGE2 and UMTS3 also implement different convolutional and turbo coding schemes (Arslan *et al.*, 2001). LTE/LTE-A adopt 1/3 code rate Turbo code because of their powerful error correcting capability and flexibility in terms of different block lengths and code rates (3GPP-TS-36.212, 2010). For burst error correction capability, Reed-Solomon codes are widely used in many applications/storage system such as computer memory storage, compact discs (CD) and digital versatile discs (DVD (Chang *et al.*, 2001; Song *et al.*, 2000).

1.3.2.2 Erasure Correction Coding

Erasure-correction is another technique for reliable communication through extra protection from redundant packets. This technique operates on packet level at the upper layer of the physical layer. An erasure code is a forward error correction (FEC) code for the binary erasure channel. Generally, for K original packets, N packets will be generated by an erasure-correction coding scheme and transmitted to the destination over single route or multiple routes. The destination can recover all the original packets once it receives a sufficient number of coded packets. In this way, erasure-correction codes establish relationships across packets to provide additional protection. Tornado codes, Low-density parity-check codes, are near optimal erasure code. However, the drawback of these codes is they that are fixed in rate. The transmitter must estimate the erasure probability, f and set the code rate $R = K/N$ before transmission.

1.3.2.3 Fountain Coding

The traditional codes such as LDPC codes and RS codes have fixed code rates. They have to estimate the code rate based on channel condition before transmission. Therefore, these coding algorithms are nearly useless if the estimated rate cannot be estimated correctly or can be changed during the transmission. In contrast to traditional fixed rate code, fountain code is a class of erasure codes with the property that a potentially limitless packets can be generated from a given set of source message. Since the number of encoded packets from the source message can be limitless, it is also known as rateless erasure code. Using fountain codes, a source potentially generate an unlimited number of coded packets and transmits the packets until the destination nodes receive a sufficient number of coded packets and decode the coded packets. The destination can decode the source message from any set of the encoded data with an efficient decoding process. Therefore the transmission reliability can be assured without requiring channel state information.

The concept of fountain codes was first presented by (Byers *et al.*, 1998/10/). Although fountain codes were originally designed for erasure channels, a lot of effort has been dedicated to their extension to general binary symmetric channels (BSC) (Palanki & Yedidia, 2004), additive white gaussian noise (AWGN) channels (Etesami & Shokrollahi, 2006), and fading channels (Castura & Mao, 2007) . The Luby transform (LT) code (Luby, 2002) is broadly viewed as the first realization of rateless codes. LT code shows excellent performance in binary erasure channel (Luby, 2002) but exhibits error floor in fading channels and impractical decoding complexity for long block codes (Palanki & Yedidia, 2004). Raptor code presented by Shokrollahi et al is a concatenation of the outer code with LT code to combat the problem of error floor in fading channels and to provide linear time encoding and decoding complexity. Raptor code has demonstrated the capacity approaching property for the binary erasure channel (Shokrollahi, 2006). Raptor code has not only beat LT code but also has shown near optimal performance on wide variety of channels. Raptor code has also provided amazing performance in AWGN channel (Etesami & Shokrollahi, 2006; Palanki & Yedidia, 2004), and Rayleigh fading channels (Castura & Mao, 2007). In general, there are two categories of physical-layer rateless codes for wireless channels: systematic (Shokrollahi, 2006; Ngo *et al.*, 2010) and nonsystematic (Etesami & Shokrollahi, 2006; Cheng *et al.*, 2009). For systematic rateless codes, the input message symbols are first transmitted to the receiver, followed by a number of encoded symbols. For nonsystematic rateless codes, only encoded symbols are broadcasted and the output does not contain input symbols. The receiver uses the classic belief propagation (BP) algorithm for decoding.

Raptor codes have been included in several mobile TV and broadband IPTV standards, e.g., DVB-H, DVB-SH, 3GPP multimedia broadcast/multicast service (MBMS), DVB IPTV, and Open IPTV forum (Mignone *et al.*, 2011).

1.3.3 Hybrid ARQ

Based on the underlying coding schemes, ARQ systems can be classified into pure ARQ systems and hybrid ARQ systems. The pure ARQ systems use a high rate block code for error detection. This type of system is simple to implement and offers high reliability in low bit error rate (BER) channel. However, in high BER channel and the pure ARQ results in a low throughput due to the large number of retransmissions. The hybrid ARQ uses a combination of FEC and pure ARQ mechanism for efficient data transmission. There are two major types of hybrid ARQ systems. The simplest version of HARQ, type I HARQ, includes parity bits for both error-correction and error-detection in each transmitted packet. After receiving the packet, the receiver first decodes the error-correction code. If the channel quality is good enough, all transmission errors should be correctable, and the receiver can obtain the original packet. If the channel quality is bad, and not all transmission errors can be corrected, the receiver will detect this situation using the error-detection code, then the received packet is rejected and a re-transmission is requested by the receiver, similar to ARQ (Costello, 2004).

The type II hybrid ARQ increases reliability by retransmitting incremental redundancy packets only when required (Mandelbaum, 1974; Kallel & Haccoun, 1990). In this type of ARQ system, parity bits for error-detection are sent on the first transmission. If errors are detected in the first transmitted packet, the second transmission will contain FEC parities and error detection. Ideally, the code rate is selected according to the actual channel state. Since in type II HARQ, coding rate can be adjusted according to channel state, it is a more efficient ARQ strategy compared to pure ARQ, type I ARQ and chase combining based ARQ schemes (Stanojev *et al.*, 2009). However under certain channel conditions type II HARQ may also require a large number of retransmission that results in unacceptable delay. To minimize delays, limiting the number of retransmission namely, truncated ARQ protocols have gained much attention in the literature (Lugand *et al.*, 1989; Yang & Bhargava, 1993; Malkamaki & Leib, 2000). A type-II hybrid ARQ scheme with a finite receiver buffer using rate 1/2 convolutional code over a two-state Markov channel was analyzed by (Lugand *et al.*, 1989). A truncated type-II hybrid ARQ scheme using block codes and only one retransmission was studied in (Yang & Bhargava, 1993) over an ideally interleaved Nakagami fading channel. In (Malkamaki & Leib, 2000), the author studied the performance of type-II hybrid ARQ schemes with a limited number of retransmissions with noisy feedback over block fading channels. They also compare the performance of type-II hybrid ARQ and FEC in terms of the reliability.

1.3.4 Multipath Forwarding

FEC improves the reliability without retransmission. Since in FEC scheme original bits and redundant bits are put in the same packets. Therefore, if one of the nodes in the route fails, the packets cannot be successfully delivered. Hence, in multihop transmission FEC scheme is not robust enough. Due to the drawbacks of ARQ-based and FEC-based schemes, researchers have proposed other schemes to improve the robustness of wireless networks. One such scheme to improve the performance of the system is multi-path forwarding or muti-path routing (Shah & Rabaey, 2002; Ganesan *et al.*, 2001//). Multi-path routing uses redundant packets through multiple paths to improve packet delivery ratio. Due to limited transmission range of wireless sensor nodes, multiple hops are usually needed for a node to send the data to the sink node. Designing suitable routing protocols to fulfill different performance demands of various applications is considered as an important issue in wireless sensor networking. There are mainly two types of routing techniques, single path routing and multiple path routing. In single path routing when the active path fails to transmit data packets, an alternative path is required to continue data transmission that may cause extra overhead and delay in data delivery. On the other hand multi path routing selects multiple paths to deliver data from source to destination and reduce end-to-end delay and congestion (Li *et al.*, 2014), increase reliability and provide robustness. Thus, multipath routing plays an important role in WSNs and many multipath routing protocols have been proposed in the literature. In (Hassanein & Luo, 2006), the authors proposed a query-based routing protocol based on the concept of multipath routing to provide path failure protection. In (Yang *et al.*, 2012), the authors proposed a network coding-based AOMDV (NC-AOMDV) routing algorithm in MANET in order to increase the reliability of data transmission and for load balancing.

CHAPTER 2

FOUNTAIN CODED COOPERATIVE COMMUNICATIONS FOR RELIABLE INTER CLUSTER COMMUNICATIONS

2.1 Introduction

Machine to machine communication (MTC) has emerged as a new communication paradigm to support a variety of applications of Internet of Things. Smart metering, telehealth, tracking and tracing, remote maintenance and control, manufacturing, facility management and security are a few possible applications in which MTC may play a leading role to improve the quality of people's lives. The M2M communication uses Long term evolution advanced (LTE-A) as backhaul network and establishes a seamless network between MTC-devices and -applications. LTE for machine type communication (LTE-M) is an extension of LTE, especially for M2M devices. In LTE-M networks, a large number of M2M devices is deployed in the field with a single eNB controlling them. According to the recent report of (Machina-Research, 2015), the total number of M2M connections will grow from 5 billion in 2014 to 27 billion in 2024. If all devices start accessing eNB then it will lead to congestion as M2M devices has limited portion of resource blocks from LTE network. One of the most effective techniques to accommodate a large volume of devices (nodes) in LTE-A network is clustering where MTC nodes are grouped into number of clusters and some portion of devices are chosen as CHs those are only allowed to access the LTE-A network. The non CH nodes in each cluster send their data to the respective CH and inter cluster traffics are routed and forwarded through the CHs to the eNB.

Nevertheless, the performance of data transmission between two neighbour CHs may degrade due to the movement and variation of the wireless channel. When a receiving CH fails to decode a packet it sends retransmission request to immediate source CH. In multi-hop network frequent retransmission request increase significant end-to end delay. Hence CH-CH communication is a bottleneck to provide reliable communication in multi-hop M2M network. In this chapter we focus on reliable inter-cluster communications. We employ intermediate non-CH nodes as relays between neighboring CHs and propose fountain coded cooperative protocols for reliable inter cluster communication between two neighboring CHs. As the number of total relays plays a significant role in cooperative communications, we first focus on relay selection procedure. In rateless coded system, the performance is improved as more relay nodes get involved. However, employing more relays between the source and the destination pair causes more signaling overhead and higher power consumption. In this regard, we focus on relay selection between two neighboring CHs and design a rateless-coded-incremental-relay selection (RCIRS) algorithm based on greedy techniques to guarantee the required data rate with a minimum cost. After that, we develop both source-feedback and non-source-feedback based fountain coded cooperative communication protocols to facilitate the data transmission between two neighbor CHs. Numerical results are presented to demonstrate the performance of these protocols with different relay selection methods over Rayleigh fading channels. It shows that the proposed source-feedback based protocol outperforms its non-source-feedback counterpart in terms of a variety of metrics.

2.1.1 Clustered M2M Network

In many applications, particularly for sensing or surveillance purpose, an M2M network might possess hundreds or thousands of nodes (devices) over a small or medium area (Karim *et al.*, 2014,; Y. Zhang & Gjessing, 2012). When many nodes are directly connected to a sink node, network complexity is significantly increased. Node clustering has been proven as a promising solution in congestion control and reliable data transmission (Huq *et al.*, 2015; Si *et al.*, 2015; Jung *et al.*, 2010). Moreover, clustering reduces energy consumption and increases the life time of the network. In this architecture, the nodes are divided into a number of clusters and some nodes are chosen as CHs which are sometimes also referred as gateways (Scaglione *et al.*, 2006). The non CH nodes in each cluster send their data to the respective CH and CHs forward the data to the sink node(e.g, eNodeB) thereby forming a one-level clustered hierarchy as shown in Figure 2.1. The following advantages can be achieved ny node clustering:

- It minimizes the routing table stored in each MTC node as data are only routed through CHs;

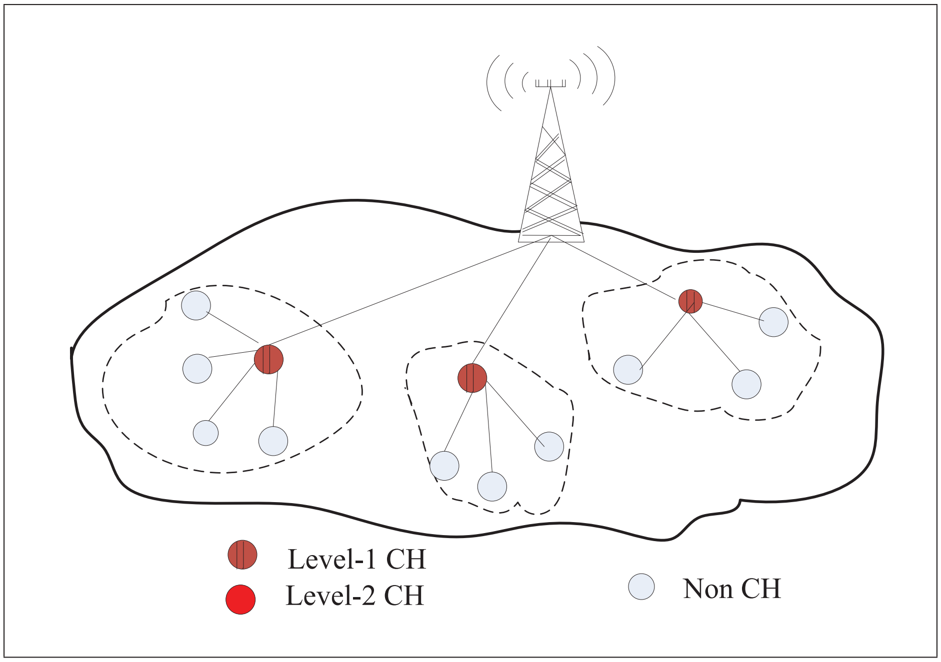


Figure 2.1 Single-hop one-level hierarchy architecture of M2M networks

- Since non-overlapping and non-neighbor clusters may use same frequency or code set for transmission, it offers good scalability and spatial reuse of resources;
- It avoids exchanging redundant messages and thus save the communication bandwidth and energy consumption;
- It offers load balancing by rotating CH role among the nodes;
- It reduces the total energy consumption of the network as the nodes can switch to the low power sleep mode by following the clustering schedules of their clusters;
- It reduces the number of access request to the base station since only the CHs are allowed to access the base station. Since CHs aggregates the data of the non CHs, the amount of data that is actually transmitted to the base station is reduced.

2.1.2 Requirements of Hierarchy Clustering

According to node configuration the clustered M2M network can be classified into homogeneous and heterogeneous clustered network. In homogeneous clustered networks, CH expires before other nodes due to the long range transmissions to the remote base station, and processing for data aggregation and protocol co-ordination. To ensure uniform drainage in this networks, the role of CH is rotated randomly and periodically over all the nodes (Heinzelman *et al.*, 2002). However the disadvantage of using a homogeneous network is that all the nodes in this network should be capable to play the CH role and therefore should possess the necessary hardware capabilities. On the other hand in heterogenous clustered network, more complex hardware and extra battery energy are embedded in few cluster head nodes to reduce the hardware cost of the rest of the network. Since only few nodes are capable to be the CH, CHs are fixed in a sense and role rotation is no longer possible. In one-level hierarchy clustering, CHs that are farthest from the sink always spend more energy than CHs those are closer to the sink. Therefore, its benefit is still limited when networks grow to thousands or tens of thousands of nodes.

In order to manage huge number of devices in a efficient way cluster based routing has been proposed in some research works (Park *et al.*, 2015; Nessa *et al.*, 2016) where all the nodes including cluster heads and sink node are related with each other in a child-parent relations as shown in Figure 2.2. In a multi-level clustering structure, data generated from terminal nodes that have no child nodes are transmitted to the cluster heads followed by data aggregation at the cluster heads for subsequent transmission to the sink node, or directly transmitted to the sink node. Since in this structure data aggregation can be performed at each level, it reduces the total amount of traffic in the network and thus improve the energy efficiency of the network in order to maximize the lifetime. Another reasons for using multi-hop routing are to extend the range of a network and provide scalable routing solution .

However, in many deployment scenarios, some machines are allowed to move and change their location in the deployment area with very low mobility, such as surveillance applications, ani-

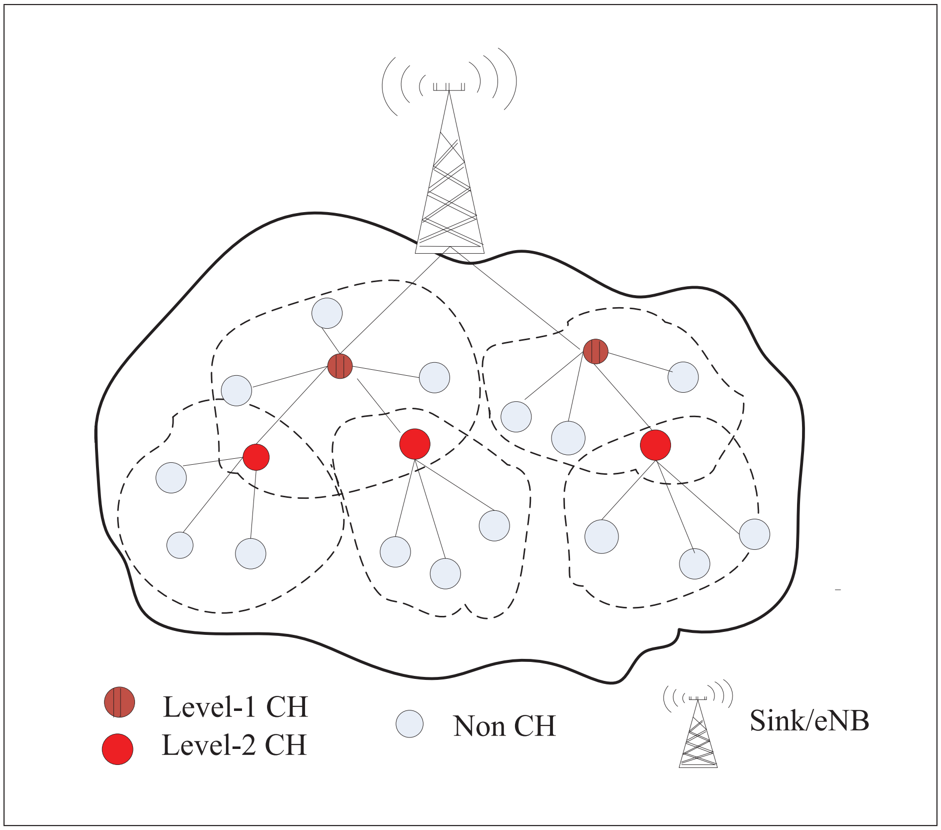


Figure 2.2   Multi-hop hierarchy architecture of M2M networks

mals in habitat monitoring applications, and traffic monitoring applications (Zhu *et al.*, 2011; Ekici *et al.*, 2006). As a consequence, when the neighboring CHs move apart from each other because of the mobility of the nodes, the performance of the data transmission may deteriorate below the threshold level. When the receiving CH fails to decode a packet successfully, it generates a retransmission request to its upstream source CH. However, with retransmission, overall end to end delay may exceed the maximum allowable delay budget, which results in packet failures in a multihop network. Therefore, CH needs to be reselected in such cases. However, frequent re-selection of CHs results in counter effect on routing and reconfiguration of resource allocation associated with CH-dependent protocols.

2.1.3 Contributions and Outline

We identify this problem as a physical layer problem and as a solution we propose to employ cooperative communications between neighboring clusters. In open literature, the work related to cooperative communications (Li *et al.*, 2012; Cui *et al.*, 2004; Madan *et al.*, 2008; Zhou *et al.*, 2008; Dohler *et al.*, 2006; Ochiai *et al.*, 2005a; Castura & Mao, 2007) mainly focuses on the performance analysis and theoretical bound for certain specific models. There has been little research done on a full and practical solution for heterogeneous M2M networks. We identify the CH-to-CH channel as a bottleneck in heterogeneous M2M network, and present a full solution to support fountain coded cooperative communications. The proposed solution consists of several schemes from relay selection to cooperative strategy, while taking into account user's QoS requirements. The main contributions of this chapter are the following:

- In most of the existing works, the number of cooperating nodes is either fixed (Cui *et al.*, 2004) or random (Zhou *et al.*, 2008; Ochiai *et al.*, 2005a; Madan *et al.*, 2008) depending on the channel and noise realizations. This work selects the cooperative nodes according to the data rate requirement. The proposed RCIRS algorithm takes into account the link between source CH to relay candidate and destination CH to relay candidate. Then it judiciously selects a minimum number of nodes while achieving the required data rate with much less computational complexity than BFS;
- In this work, we also develop two communication protocols based on fountain coding, namely source-feedback and non-source-feedback cooperative communication protocols to improve the performance of data transmission between two neighboring CHs. We evaluate the performance of our proposed protocols with different relay selection methods using the metric of transmission efficiency rather than the outage probability. This strategy is more realistic since in rateless coded systems, the outage probability is always driven to zero if there is no restriction on decoding delay.

The rest of the chapter is organized as follows. We first review the existing work in Section 2.2 and introduce our system model of cooperative communications over clustered M2M networks in Section 2.3. We then present the relay selection algorithm and fountain coded cooperative communication protocols in Section 2.4 and Section 2.5, respectively. Finally, Section 2.6 demonstrates the simulation results and Section 2.7 concludes this chapter.

2.2 Related Work

Cooperative communication is an effective way to combat the effects of channel fading and improve the throughput, power efficiency and coverage in wireless networks (Laneman *et al.*, 2004; Zhang & Zhang, 2013; Li *et al.*, 2012). In wireless networks when the source transmits to the destination, neighboring nodes can overhear the transmission and are able to transmit to the same destination. Cooperative communications utilize the broadcast nature of wireless channel by considering the neighboring nodes as relays and allow them to transmit the overheard information to the destination. Destination thus receives multiple replicas of the signals from independent fading paths and achieves diversity even though it is equipped with a single antenna (Laneman *et al.*, 2004). Currently, the researchers are giving more emphasis to explore cooperative strategies and protocols to achieve higher performance and reliability in communications (Laneman *et al.*, 2004; Li *et al.*, 2012; Ochiai *et al.*, 2005a; Cui *et al.*, 2004; Madan *et al.*, 2008). Distributed beamforming schemes have been proposed in (Sendonaris *et al.*, 2003) and (Ochiai *et al.*, 2005b), where the source and cooperative relays need to be synchronized and adjust the phase of their transmissions such that their signals can coherently be added at the destination. Distributed beamforming require significant modification to existing radio frequency (RF) front ends, which will increase system complexity and costs. Some research works also focused on designing distributed spacetime coding for cooperative systems (Seddik *et al.*, 2008; Zhou *et al.*, 2008; Dohler *et al.*, 2006), where a number of nodes transmit the different columns of a space-time coding matrix simultaneously to the destination. In (Zhou *et al.*, 2008; Dohler *et al.*, 2006), the authors analyzed cooperative communication based on distributed space-time block coding (STBC) for wireless sensor network and showed that cooperative communication is more energy efficient than non-cooperative direct communication. However, distributed STBC requires strict synchronization among transmitters, which is difficult and even impossible in most mobile cases. Previous works (Zhou *et al.*, 2008; Dohler *et al.*, 2006) addressed the fixed rate code. The outage probability in their works cannot reach zero if the precise channel state information (CSI) is unavailable at the transmitter.

Among many other channel coding techniques, fountain code - a channel coding technique (Luby, 2002; Etesami & Shokrollahi, 2006; Etesami *et al.*, 2004) that adapts its rate according to the variation of channel condition, has attracted a lot of research interest. For its capacity approaching potentiality and rateless property, it appears as a promising solution for data communication in many wireless communication systems including relay networks (Castura & Mao, 2007; Nessa *et al.*, 2012) and cognitive radio systems (Chaoub & Ibn-Elhaj, 2014). As a promising feature, it does not require any synchronization mechanism like virtual multiple-input-multiple-output (MIMO). Also it does not require CSI at the transmitter as fixed rate code.

In open literature, there exists several works investigating the performance of rateless codes over wireless relay channel. The first rateless coding framework over relay channels was presented by (Castura & Mao, 2007), where the relay assists the source as a secondary antenna once it decodes source information successfully. In this framework, the relay node synchronizes itself with the source before starting transmission. The source and relay then transmit data to the destination using space time Alamouti code (Alamouti, 1998). In (Liu & Lim, 2009), the authors proposed several single relay cooperative schemes and derived their achievable rate in flat Rayleigh fading channel. In(Yang & Host-Madsen, 2006), the authors studied the performance of fountain codes in low power regimes. The performance of fountain codes in wireless body area network with respect to reliability and energy consumption is studied in (Abouei *et al.*, 2011). In (Nikjah & Beaulieu, 2011), the authors proposed several single parameter relay selection protocols based on information accumulation and rateless coding schemes. There, relay nodes send acknowledgement to the destination after successfully decoding the source information, and the destination then finds out the best relay to transmit information to the destination. The performance of fountain code in the presence of multiple relay nodes under block fading Rayleigh channel was studied in (Molisch *et al.*, 2007). The authors showed that information accumulation consumes less energy and requires less transmission time than that of energy accumulation.

In cooperative communications, a relay node is located between the source and the destination and assists the communication in a two hop manner to archive higher data rate than the data rate of direct link. In the literature, comprehensive studies have been given on relay selection in the cooperative communications with fixed code rate (Zhao *et al.*, 2006; Bletsas *et al.*, 2006; Onat *et al.*, 2008; Bali *et al.*, 2010). While several relay selection schemes are proposed for both decode and forward(DF) and amplify-and-forward (AF) networks, most of these schemes are on single relay selection. A distributed scheme for the selection of the single relay node in the presence of multiple relays based on instantaneous channel condition was presented by (Bletsas *et al.*, 2006). It was shown for a single source-destination pair, the selection of the best relay node among multiple relay nodes can offer same level of performance as employing many ordinary relay nodes (Zhao *et al.*, 2006). In (Onat *et al.*, 2008), the authors proposed threshold-based selective relaying schemes in order to minimize the end-to end bit error rate (BER) in cooperative decode and forward system. A distributed scheme for multiple relay selection based on source and relay channel and threshold optimization was proposed by (Bali *et al.*, 2010). They assumed that all the relays have same average SNRs to the source and optimized the threshold value on this assumption. However, in practical, the relay nodes are located randomly between the source and the destination and experienced SNRs at relay nodes vary according to the distance and fading of the wireless channel.

Most existing works on fixed code rate system address energy combination of orthogonal transmissions from different paths at the receiver. In rateless coded system, however, receiver collects mutual information from different independent paths to decode information. Therefore, the relay selection scheme for fixed rate cannot provide the optimal throughput when applied to rateless coded cooperation. In fact, the investigation on the relay selection has not received much attention yet regarding the cooperative communications using rateless code. In (Molisch *et al.*, 2007), as soon as relay nodes decode the message, they start transmitting to the destination with the possibility of having a poor link. The existing works in (Bletsas *et al.*, 2006; Nikjah & Beaulieu, 2011; Zhang *et al.*, 2010) mainly focus on how to select the best relay in the presence of multiple relay for a single source-destination pair so that the highest transmission rate can be achieved (Nikjah & Beaulieu, 2011; Zhang *et al.*, 2010). In (Nikjah & Beaulieu, 2011), the destination selects the best relay from a set of nodes after decoding the source information. The achievable data rate with the cooperation of a single relay may not be able to attain the target level. Moreover, in wireless M2M network, single relay selection may not be robust enough to provide reliability due to the link instability and resource limitation of the MTC nodes. Multiple relay selection utilizing the advantage of fountain codes becomes a critical challenge in such a research context that we solve in our work.

2.3 System Model

In this work we investigate LTE-A connected heterogeneous M2M network where the network consists of different types of devcie/nodes with respect to battery supply and functionality. The M2M domain is then further partitioned into a number of overlapping and disjoint clusters and some powerful devices among the devices are selected as cluster heads (CHs) as illustrated in Figure 2.3. In our setup, CH emulates the role of the base stations, coordinates transmissions within the cluster, collects data, communicates with neighboring CHs and forwards data to the LTE-A eNB using single hop or multihop communications. We also assume that nodes are equipped with both a long range and a short range interface. Nodes use short range radio interface to create an ad-hoc capillary network among the nodes in local M2M domain and long range interface to connect to LTE-A network. The reason behind clustering is to avoid congestion and increase the efficiency of data transmission by allowing limited number of nodes to access eNB.

Evidently the level of congestion control depends on the number of CHs. On the other hand, accommodating a large number of nodes in a cluster may also cause congestion inside a cluster and increase the average distance between cluster heads and the group members that will in-

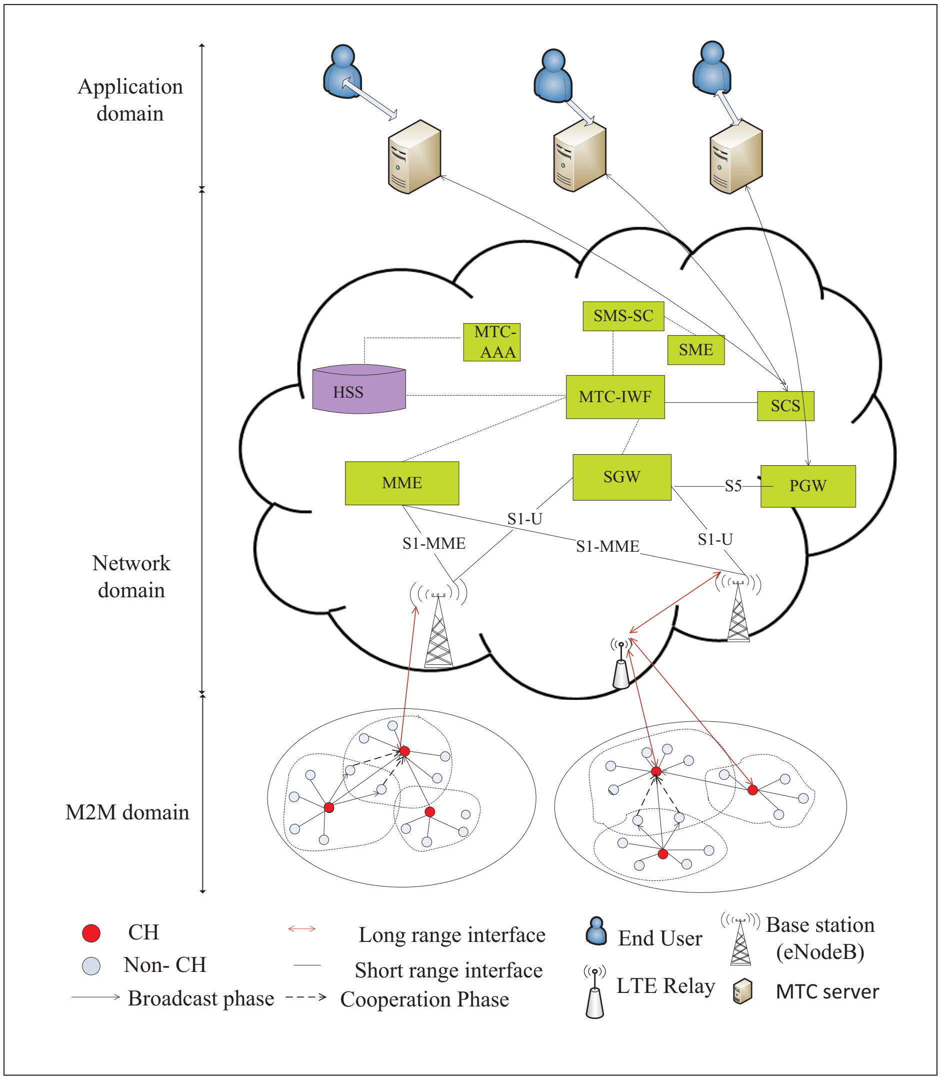


Figure 2.3 An LTE-A connected heterogeneous M2M networking architecture

crease energy consumption for the intra communication. Hence, there is a tradeoff between the number of CHs and the size of the CHs. Wireless sensor network (WSN) are similar to M2M communication in characteristic features to a certain extent. In the literature, many clustering algorithm have already been proposed in WSN fields, which can be correlated with clustering in M2M communication. Communication capability, communication link quality, storage status and battery status of each nodes are mainly considered during CH selection (Huq *et al.*, 2015; Jung *et al.*, 2010; ExaltedProject, 2011). However, cluster design and CH selection are

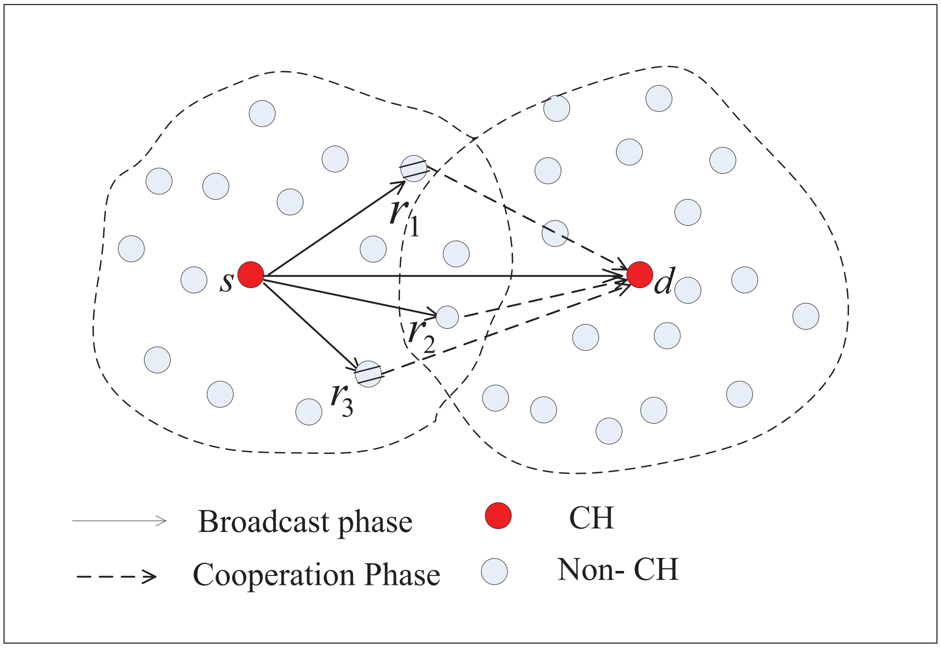


Figure 2.4 Cooperative communications framework over clustered M2M network

out of the scope of this work. In this work we only focus on the reliable data transmission between two clusters. The assumptions of the system model is as follows:

- M2M devices are randomly deployed in the cell with different radio range, functionality and processing capabilities,
- According to processing capabilities, functionality and battery supply there are three types of nodes in the M2M domain. CHs are more powerful nodes in terms of battery supply and processing capabilities and equipped with both long and short range interfaces,
- There are some ordinary nodes those have simple processing capabilities and only have short range communication capabilities,
- There are also some nodes those have more processing capabilities than the ordinary nodes and equipped with short rang communication capabilities. This type of nodes can relay the traffic between two neighboring CHs,
- Devices are clustered appropriately.

Table 2.1 Notation

Symbol	Definition
H	Number of bits in relay_probe message
C_{target}	Target end to end data rate of s-d pair
C_{sd}	Achievable data rate for s-d pair
C_{sr_i}	Achievable data rate for s-r_i pair
C_{r_id}	Achievable data rate for of r_i-d pair
C_{cdr}	Achievable cooperative data rate of the network
γ_{ab}	Channel gain of a-b pair
τ_{ab}	Time taken by b to decode *relay_probe* message transmitted by a
P_0	Transmission power of each node
$\mathscr{C}$	Set of initial relay candidates
$\mathscr{A}_x$	Set of successful relay candidates
$\mathscr{A}_{min}$	Set of selected relay nodes
N	Number of initial candidates in set $\mathscr{C}$
R	Number of relays in set $\mathscr{A}_{min}$
ζ	The transmission efficiency
$\Omega(x)$	Degree distribution of Raptor code in noisy channels
Z_{c_o}	Channel log likelihood ratio
$G_{a,b}$	Path loss between node a and node b

Base on the aforementioned assumption, in Figure 2.4, we present an example system model that consists of two overlapping clusters with many edge nodes that can sense signal from both CHs. This figure illustrates a simple scenario of data transmission between two overlapping clusters where source CH, s transmits data targeting destination CH, d. Exploiting the intrinsic nature of wireless medium, the neighbor nodes r_1, r_2, r_3 overhear the transmission between two CHs and are able to facilitate the transmission by forwarding the received information to the neighbor CHs. Correspondingly, r_1, r_2, r_3 act as relays between s and d, and thus d achieves diversity by receiving signals from different fading paths.

We assume, the transmit power of each transmit node s is restricted to P_s and the noise at each receive node j is assumed to be white Gaussian with a variance of σ_j^2. Denoting the transmitted symbols of s as X, the received symbols at r_i and d can be expressed as

$$Y_j = H_{sj}X + W_j, \tag{2.1}$$

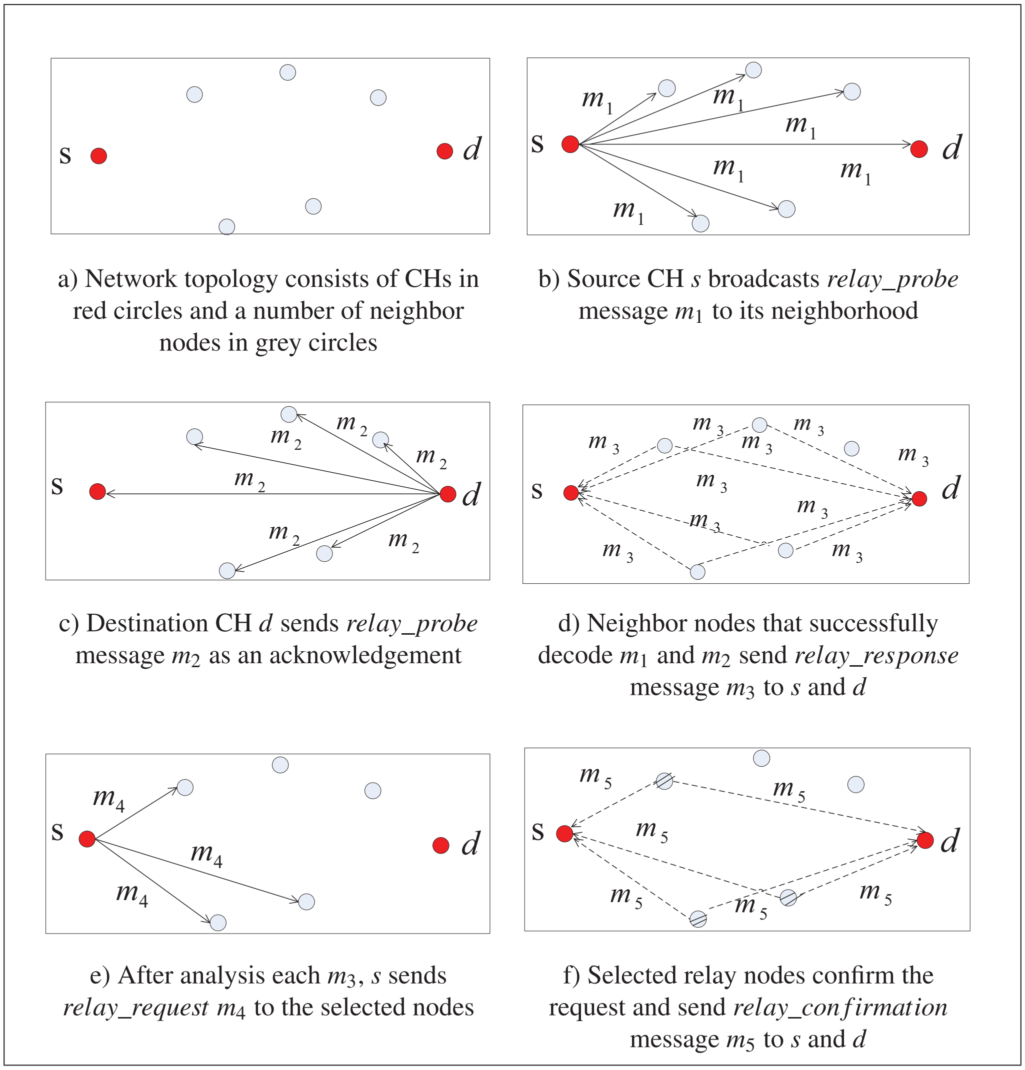
where $j = \{r_i, d\}$; W_j is the Gaussian noise at j with variance σ_j^2; H_{sj} is modeled as $\sqrt{G_{sj}}h_{sj}$ where $h_{sj} \sim \mathscr{CN}(0,1)$ which is a circularly symmetric complex Gaussian random variable with zero mean and unit variance, and G_{sj} denotes the path loss between node s and node j. Then the received SNR at j for a pair s-j can be expressed as

$$\gamma_{sj} = H_{sj}^2 P_s / \sigma_j^2. \tag{2.2}$$

In order to avoid interference, we assume that orthogonal channels (Zhu & Wang, 2009, 2012) are allocated to different terminals i.e., inter-user orthogonality. The source transmits to the destination and relays through one channel and the relays transmit to the destination through other different orthogonal channels. This assumption can be easily implemented in practice using orthogonal frequency division multiplexing (OFDM) modulated WiFi technology, where different sub-carriers are assigned to different users. After relay selection, we assume that all the nodes use their own fountain coding routine to encode the data. Table 2.1 lists the notation used in this chapter. In the subsequent sections, we will use source node (destination node) and source CH (destination CH), interchangeably.

2.4 Rateless Coded Relay Selection

The selection of the relay nodes plays a critical role to improve the performance of the cooperative system (Liu & Lim, 2009; Bletsas *et al.*, 2006; Zhang *et al.*, 2010). However, involving more relay nodes between source and destination impose more signalling overhead and power consumption. Considering the cost and performance issues in M2M network, we propose a relay selection algorithm called rateless coded incremental relay selection (RCIRS), aiming at achieving the given target data rate by employing minimum number of relay nodes. We consider a decode and forward relaying system where source CH transmits its information to the



a) Network topology consists of CHs in red circles and a number of neighbor nodes in grey circles

b) Source CH s broadcasts *relay_probe* message m_1 to its neighborhood

c) Destination CH d sends *relay_probe* message m_2 as an acknowledgement

d) Neighbor nodes that successfully decode m_1 and m_2 send *relay_response* message m_3 to s and d

e) After analysis each m_3, s sends *relay_request* m_4 to the selected nodes

f) Selected relay nodes confirm the request and send *relay_confirmation* message m_5 to s and d

Figure 2.5 Pictorial representation of relay selection procedure

destination CH and relays using fountain codes. After successfully decoding the source CH's information, relay nodes encode the received information using fountain codes. The relay nodes and the source node then continue their transmission to the destination until the source information is successfully decoded at the destination.

2.4.1 Relay Selection Procedure

We illustrate our proposed relay selection procedure in Figure 2.5. Figure 2.4 a) shows an example network topology with a source CH and a destination CH by red colored circles, and a number of neighbor nodes by grey colored circles. In the beginning of relay selection, source CH, s broadcasts *relay_probe* message denoted by m_1 as shown in Figure 2.4 b) using predefined rateless code to its neighbor nodes and continues its broadcasting until it receives an acknowledgement from destination CH, d. Let I bits of data to be broadcasted as *relay_probe* message. We assume I is a long enough number. The addresses of the source and destination CHs are included in the *relay_probe* message. After successfully decoding of m_1, d sends a *relay_probe* message denoted by m_2 as an acknowledgment to s using rateless code which is shown in Figure 2.4 c). Then s estimates the end to end data rate for s-d pair based on the time it takes for the successful decoding of *relay_probe* message transmitted by d. This *relay_probe* message transmitted by d is reliably decoded at s when

$$\tau_0 C_{sd} = \tau_0 log_2(1+\gamma_{sd}) \geq I \tag{2.3}$$

where τ_0 is the duration of transmission and γ_{sd} is the received SNR at s for s-d pair. In ideal fountain code, the receiver is capable to recover the source information as soon as the accumulated mutual information equal the entropy of the codeword. However it is impossible to generate universal fountain code that is simultaneously perfect at all possible rates. In practical fountain code, the receiver is capable to recover the source information with overhead that is not too large (Etesami *et al.*, 2004). An overhead like this will only lead to a loss of rate and does not impact much on the analysis here. In order to simplify the analysis, ideal fountain codes and decoders are assumed at the receivers in this section. Therefore, the achievable data rate per unit bandwidth for s-d pair is given by

$$C_{sd} = log_2(1+\gamma_{sd}) = I/(\tau_0). \tag{2.4}$$

At the same time, neighbor nodes process the received *relay_probe* messages from different CHs and measure the achievable data rate. Let assume neighboring node r_i takes τ_{sr_i} and τ_{dr_i}

time to decode *relay_probe* messages of s and d, respectively. Hence, the achievable data rate for s-r_i pair is

$$C_{sr_i} = log_2(1+\gamma_{sr_i}) = I/\tau_{sr_i}, \tag{2.5}$$

where, γ_{sr_i} the received SNR at r_i. Similarly, the achievable data rate for r_i-d pair is given by

$$C_{r_id} = log_2(1+\gamma_{r_id}) = I/\tau_{r_id}, \tag{2.6}$$

where, γ_{r_id} is the received SNR at r_i for r_i-d pair. Among the neighbor nodes, only those who receive *relay_probe* messages from both source and destination CHs respond to the *relay_probe* with *relay_response*, which is denoted by m_3 as shown in Figure 2.4 d). The *relay_response* message includes the data rates C_{sr_i} and C_{r_id}. Source CH, s then analyzes each *relay_response* message and takes only those nodes whose source to relay data rate, C_{sr_i} are greater than C_{sd}. We refer these nodes as initial relay candidates and the set of candidate relays as initial candidate set $\mathscr{C}$.

The next step is to search for a subset $\mathscr{A}_{min}$ from the initial candidate set $\mathscr{C}$. Here, $\mathscr{A}_{min}$ denotes our desired set of relay nodes that consists of minimum number of relay nodes and contributes to meet the target CH-to-CH data rate. In regards to this, we propose a search criterion and a greedy algorithm, namely RCIRS algorithm. In the next two subsections, we explain the proposed search criterion and RCIRS algorithm. After the selection of set $\mathscr{A}_{min}$, the source sends the relay request message m_4 to the selected nodes as shown in Figure 2.4 e). Then the selected nodes confirm this request and send a *relay_confirmation* message, which is denoted as m_5, to both s and d as shown in Figure 2.4 f). Thus, during the design phase of the network, source CH judiciously selects the minimum number of relay nodes from a large set of neighbor nodes. When the network starts to actively operate, the source node sends data only targeting the selected relay nodes along with the destination node. Then the selected nodes forward the source data only to the destination and thus the overall cost is reduced while the target data rate is maintained.

2.4.2 Relay Selection Problem

The main goal of the relay selection problem is to find a set of relay nodes $\mathscr{A}_{min}$ from $\mathscr{C}$, that has minimum number of relay nodes and also satisfies the target data rate constraint, C_{target}. To solve this problem, we establish a search criterion. In the search criterion, we introduce a parameter, namely achievable cooperative data rate, C_{cdr} that represents the data rate between source CH and destination CH attained by the co-operative participation of source, relay and destination nodes. The search criterion is stated as below:

$$C_{cdr} \geq C_{target}, \tag{2.7}$$

where

$$C_{cdr} = \left(\frac{C_{sd} + \sum\limits_{r_i \in \mathscr{A}_x} C_{r_i d}}{1 + \sum\limits_{r_i \in \mathscr{A}_x} \frac{C_{r_i d}}{C_{sr_i}}} \right), \tag{2.8}$$

where, $\mathscr{A}_x \subseteq \mathscr{C}$ is a successful candidate set that satisfies the criterion in Equation 2.7. The relationship of the parameter C_{cdr} with previously defined data rates of s-d, s-r_i and r_i-d pair shown in Equation 2.8, is derived by using Equation 2.4, 2.5, 2.6 and the following equations,

$$\tau.C_{sd} + \sum_{r_i \in \mathscr{A}_x} (\tau - \tau_{sr_i}) C_{r_i d} = I, \tag{2.9}$$

$$\frac{I}{\tau} = \left(\frac{C_{sd} + \sum_{r_i \in \mathscr{A}_x} C_{r_i d}}{1 + \sum_{r_i \in \mathscr{A}_x} \frac{C_{r_i d}}{C_{sr_i}}} \right), \tag{2.10}$$

where, τ is the minimum time taken by the destination with the cooperation of a set of nodes $\mathscr{A}_x$ to successfully decode the source message and $\frac{I}{\tau} = C_{cdr}$. Let consider, N is the total number of relays in the initial candidate set $\mathscr{C}$. So, the number of all possible successful candidate set $\mathscr{A}_x$ could vary from 0 to $2^N - 1$. In the next paragraph, we have shown an example problem formulation of searching $\mathscr{A}_{min}$ from set $\mathscr{C}$.

Let α_i be an indicator variable which is equal to 1 if r_i is selected as relay and 0 otherwise, the minimal selection problem can be formally defined as follows:

$$\begin{aligned} & \min \sum_{i=1}^{N} \alpha_i \\ & \text{s.t.} \;\; C_{cdr} \geq C_{target}, \\ & \alpha_i \in \{0,1\}, \forall r_i \in \mathscr{C}. \end{aligned} \tag{2.11}$$

The above problem is a non linear integer optimization problem. Now if the size of the network is high i.e. if N is a large number, finding $\mathscr{A}_{min}$ from all possible $\mathscr{A}_x \subseteq \mathscr{C}$ sets by using conventional Brute-Force search would require huge computational effort. The computational complexity of BFS increases exponentially with the number of relay candidates N, that limits its application in practical systems. With N initial relay candidates in the selection process, an exponential complexity of $2^N - 1$ would be required to find out $\mathscr{A}_{min}$ set. In order to reduce the computational complexity of searching $\mathscr{A}_{min}$, we propose a greedy algorithm RCIRS in the next subsection.

2.4.3 Rateless Coded Incremental Relay Selection (RCIRS) Algorithm

Algorithm 2.1. Rateless Coded Incremental Relay Selection

Input: $\mathscr{C}, C_{target}$
Output: $\mathscr{A}_{min}, C_{cdr}$
1: $C_{cdr} \leftarrow C_{sd}, Sum\_C_{rd} \leftarrow 0, Sum\_C_{rd}/C_{sr} \leftarrow 0$
2: $\mathscr{A}_{min} \leftarrow \emptyset$
3: while $C_{cdr} < C_{target}$ do
4: for each $i \in \mathscr{C} \backslash \mathscr{A}_{min}$ do the following do
5: $\Delta_{r_i} = \left(\frac{C_{sd}+Sum\_C_{rd}+C_{r_id}}{1+Sum\_C_{rd}/C_{sr}+C_{r_id}/C_{sr_i}} \right)$
6: end for
7: $r_{i^*} \leftarrow \arg \max_{i \in \mathscr{C} \backslash \mathscr{A}_{min}} \Delta_{r_i}$, $\Delta_{r_{i^*}} = max(\Delta_{r_i})$
8: $\mathscr{A}_{min} \leftarrow \mathscr{A}_{min} \cup r_{i^*}$
9: $C_{cdr} \leftarrow \Delta_{r_{i^*}}$
10: $Sum\_C_{rd} \leftarrow Sum\_C_{rd} + C_{r_{i^*}d}$
11: $Sum\_C_{rd}/C_{sr} \leftarrow Sum\_C_{rd}/C_{sr} + \frac{C_{r_{i^*}d}}{C_{sr_{i^*}}}$
12: end while
13: return $\mathscr{A}_{min}, C_{cdr}$

In Algorithm 2.1, different steps of RCIRS algorithm is depicted. The basic idea of RCIRS algorithm is that nodes in the set $\mathscr{C}$ are included in cooperative group one by one so that the cooperative data rate is incrementally increased until the condition in Equation 2.7 is satisfied. In order to minimize the number of relays, each time the candidate that makes the maximum increment to C_{cdr} is added in the cooperative group. At the beginning, selected $\mathscr{A}_{min}$ is an empty set, since no relay has been selected, and C_{cdr} is initialized with C_{sd} by considering the signal received from only s (line 1-2). After the initializations, C_{cdr} is compared with C_{target} (line 3). If C_{cdr} is smaller than C_{target}, the achievable cooperative data rate Δ_{r_i} is computed by using each candidate relay r_i (line 4-6). Then the candidate relay that maximizes Δ_{r_i} is selected and included in the selected relay set $\mathscr{A}_{min}$ (line 7-8). C_{cdr}, Sum_C_{rd} and Sum_C_{rd}/C_{sr} are updated accordingly (line 9-11). This process is repeated for remaining elements of $\mathscr{C}$ until Equation 2.7 is satisfied (line 12). At the end, the algorithm returns the desired relay set $\mathscr{A}_{min}$ and cooperative data rate of the system C_{cdr} (line 13). In this way, the RCIRS algorithm finds a near minimal $\mathscr{A}_{min}$ by limiting the candidate search space within $N + (N-1) + \dots 1 = N(N+1)/2$ sets, which is much lower than the search space of BFS $(2^N - 1)$.

2.5 Fountain Coded Cooperative Protocols

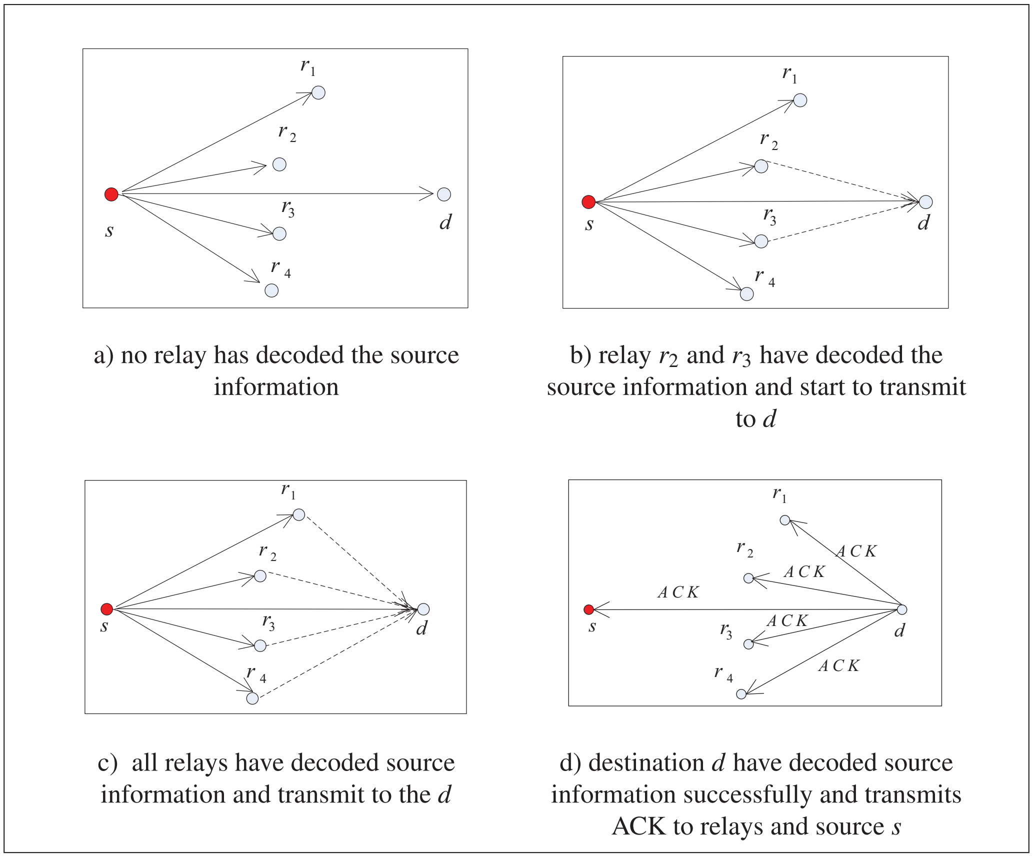
We propose two fountain coded cooperative communication protocols, namely source-feedback and non-source-feedback where source CH and relay nodes use their own fountain routines to encode information and then transmit the encoded information to the destination through different orthogonal channels. To recover the original data from the transmitted bits, the decoder must be acknowledged about the degree distribution and set of neighbors for each encoding symbol. There are so many ways to inform the decoder about this information depending on the types of applications. We assume that the decoder knows the encoding degree distribution of received encoded symbols via an extra robust direct sequence spread spectrum (DSSS) channel with long sequence length. This is different from using identical random number generator at both encoder and decoder as in (Liu & Lim, 2009), which needs strict synchronization and increases the complexity in a multiple relay network.

2.5.1 Source-feedback Based Protocol

In this protocol, a source CH encodes its information using fountain code and transmits to the destination CH and relays. Relay nodes attempt to decode source data and forward the source information to the destination CH using their own fountain codes as soon as the information is decoded successfully at the relay nodes. Source CH selects this relay nodes by appropriate relay selection algorithm. Both source CH and relays then continue their transmissions until they receive an acknowledgment from the destination CH indicating that the reception has been successful. The Source-feedback based protocol is illustrated in Figure 2.6 where the source CH s, destination CH d in red circles and a number of neighbor nodes in grey circles. The protocol works as follows:

a. In the beginning of this protocol the source transmits its encoded data targeting the destination and relays as shown in Figure 2.6(a);

b. Both relays and destination accumulate information from source transmission. Since the source to relay link is superior than source to destination link, relays decode the informa-



a) no relay has decoded the source information

b) relay r_2 and r_3 have decoded the source information and start to transmit to d

c) all relays have decoded source information and transmit to the d

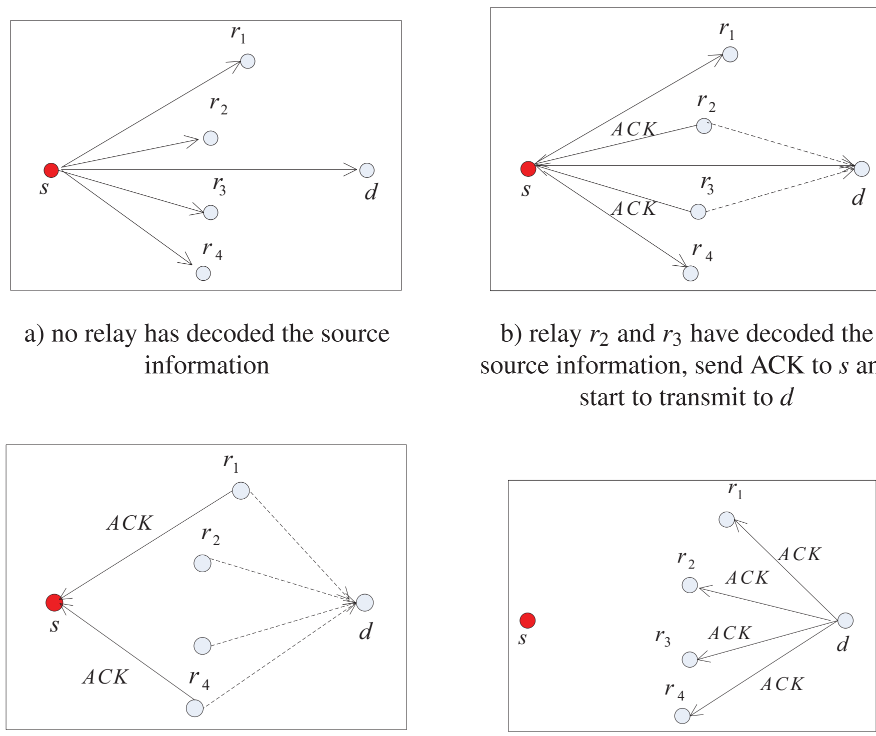
d) destination d have decoded source information successfully and transmits ACK to relays and source s

Figure 2.6 Steps of the source-feedback based protocol

tion faster than the destination. Even some of the relay nodes decode the source information earlier than other relay nodes as shown in Figure 2.6 b). In Figure 2.6 b) we see, relay r_2 and r_3 decode the source information successfully before any other relays and start to transmit to the destination. As soon as relay nodes decode the source information they co-operate with the source to transmit to the destination. Each relay uses its own fountain encoding routine to encode information and then transmit to the destination through pre-allocated orthogonal channel;

c. Finally r_1 and r_4 decode the source information successfully and join in the cooperative group to transmit the source information to the destination as shown in Figure 2.6 c). The destination collects information from both source and relays, and attempts to decode information once the accumulated mutual information is slightly greater than the source information. If the decoding is successful, then it sends an ACK targeting the source and relays as shown in Figure 2.6 d). Otherwise it continues to collect more information. This procedure continues until successful reception at the destination.

2.5.2 Non-source-feedback Based Protocol



a) no relay has decoded the source information

b) relay r_2 and r_3 have decoded the source information, send ACK to *s* and start to transmit to *d*

c) relay r_1 and r_4 have decoded the source information, send ACK to *s* to ceases its transmission and only relays transmit to the *d*

d) destination *d* have decoded source information successfully and transmits ACK to relays

Figure 2.7 Steps of the non-source-feedback based protocol

In this protocol, source CH transmits the data stream encoded by a fountain codes targeting the destination and relay nodes. The relay nodes listen to the source data; as soon as a relay acquired sufficient information to decode source data, it transmits an ACK to the source that its reception was successful. Instantly the relay node switches from reception to transmission mode and co-operates with the source to transmit to the destination. Once the source has received acknowledgments from all of the relay nodes, it ceases its transmission hoping that the destination can successfully receive the remaining information from relays. The protocol works as follows:

a. Source generates a large number of encoded symbols and transmit its encoded data targeting the destination and relays as shown in Figure 2.7 a);

b. Relays and the destination d consistently receive signals from source and accumulate the mutual information to decode the source information;

c. As soon as a relay node has sufficient information to decide on a codeword, it sends acknowledgement to the source and switches from reception mode to transmission mode. At this point, it encodes information using its own fountain encoding routine and transmit on the pre-allocated orthogonal channel. As shown in Figure 2.7 b) relay r_2 and r_3 successfully decode the source information, sends ACK to the destination and start to transmit to the destination;

d. Finally relay r_1 and r_4 decode the source information successfully and send ACK to the the source, s as shown in Figure 2.7 c). Source ceases its transmission after receiving acknowledgement from all relay nodes. The suspension of source nodes may help to decrease the energy consumption. By this time, if the destination d can not accumulate enough partial information from source transmission, it depends on the relay nodes to collect the remaining information. After successfully decoding of source information, the destination sends ACK to relays as shown in Figure 2.7 d).

2.5.3 Encoding and Decoding of Raptor Code

In our proposed relaying schemes, the source and relays use a special class of fountain code, namely Raptor code to encode their information. Raptor code is universally capacity achieving over the binary erasure channel (BEC) (Etesami & Shokrollahi, 2006) and nearly capacity-achieving over other channel models such as binary symmetric channels (BSC), AWGN and

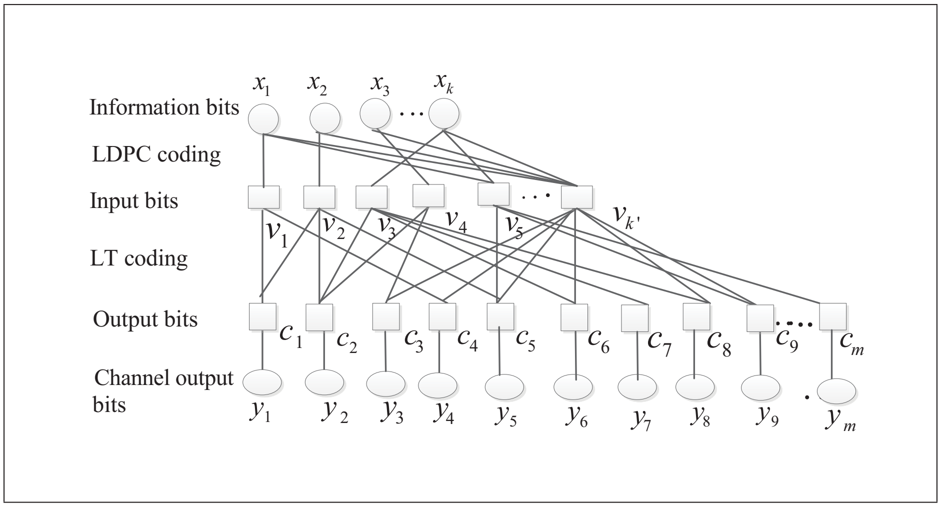


Figure 2.8 The factor graph of Raptor code. The graph is truncated to length m

fading channels (Castura & Mao, 2007; Etesami *et al.*, 2004). It is a concatenation of two codes, a pre-code or outer code and an inner code. Usually LDPC and LT codes are used as outer and inner codes, respectively. However, in most practical settings Raptor codes outperform LT-codes in every aspect.

$$\begin{aligned}\Omega(x) = 0.006x + 0.492x^2 + 0.03396x^3 + 0.2403^4 + 0.006^5 + \\ 0.096x^8 + 0.049x^{14} + 0.018x^{30} + 0.0356x^{33} + 0.033x^{200}.\end{aligned} \tag{2.12}$$

We assume source CH s wants to transmit k-symbols information block to destination CH d. In Raptor encoding, the k-symbols information $\{x = x_1, x_2, ...x_k\}$ is first encoded by the pre-code which is typically high-rate LDPC code to produce a k'-input symbols $\{v = v_1, v_2, ...v_{k'}\}$. Then LT-codes that follows the degree distribution as mentioned in Equation 2.12 (Etesami & Shokrollahi, 2006) is applied to v to create Raptor codeword $\{c = c_1, c_2, ...c_m....\}$. In our system, A rate-0.98 LDPC code is implemented as the outer code of Raptor codes. In Figure 2.8, the factor graph of Raptor code is shown that is referred as G_m and truncated at block length m where channel output $\{y = y_1, y_2, ...y_m\}$ corresponds to transmitted Raptor codeword $\{c = c_1, c_2, ...c_m\}$.

As mentioned before, the degree of the encoded information is transmitted to the destination by an extra robust CDMA channel with long sequence length. The decoder collects information progressively and makes first decoding attempts when the received accumulated information is little bit larger than the source information. In noisy channel, the decoding of raptor code is accomplished using the standard Belief Propagation (BP) algorithm. At the l^{th} decoding attempt, it performs BP decoding on factor graph G_m by iteratively passing the LLR (log-likelihood ratio) messages from input bits$\{v_1...v_{k'}\}$ to output bits $\{c_1...c_m\}$ and then from output bits back to input bits. Let $\mu^{j,l}_{c_o,v_i}$ and $\nu^{j,l}_{v_i,c_o}$ denote the message passed from the output bit c_o to the input bit v_i and input bit v_i to the output bit c_o, respectively at the j^{th} iteration of l^{th} decoding attempt. In every iteration, the following message update rules are applied in parallel to all input and output nodes in the factor graph (Etesami & Shokrollahi, 2006)

$$\tanh\frac{\mu^{(j,l)}_{c_o,v_i}}{2} = \tanh\frac{(Z_{c_o})}{2}\prod_{i'\neq i}\tanh\frac{\nu^{j,l}_{v_i,c_o}}{2}, \tag{2.13}$$

$$\nu^{(j+1,l)}_{v_i,c_o} = \sum_{o'\neq o}\mu^{(j,l)}_{c_{o'},v_i}, \tag{2.14}$$

where Z_{c_o} is log-likelihood ratios (LLR) of the transmitted bit c_o. Since we use binary phase shift keying (BPSK) as modulation scheme, the transmitted codeword $c_o \in 0,1$ is of equal probability. In Rayleigh fading channel, the channel intrinsic log likelihood information corresponding to the output node c_o while channel state information is available at the receiver, is formulated as (Hou *et al.*, 2001)

$$Z_{c_o} = \ln\left(\frac{P(y_o = 0|c_o)}{P(y_o = 1|c_o)}\right) = \frac{2}{\sigma^2}y_o.a, \tag{2.15}$$

where $\sigma^2 = \frac{P_0}{\sigma_0^2}$, y_o is the noisy observation of the channel and a is the normalized Rayleigh fading factor with $E[a^2] = 1$ and density function $f(a) = 2a\exp(-a^2)$. In the end of l^{th} decoding attempt, if the transmitted codeword is decoded correctly, the receiver sends an ACK through a noiseless feedback channel to the source to terminate the transmission of the current code word. Otherwise it collects more output symbols and attempts again to decode the codeword.

2.5.4 Transmission Efficiency

In rateless coded system, the source node generates a large number of symbols and transmits to the destination until it receives any ACK from the destination. Hence, the probability of outage is always driven to zero unless any constraint is imposed on decoding delay. In this work, transmission efficiency rather than outage probability is considered as a primary metric to evaluate the performance of the proposed protocols. In the following, we evaluate the performance of our proposed protocols based on the required time to decode the source information correctly at destination.

Let n denotes the number of time slots required for the destination to collect enough information from source and all relays before it successfully decodes a k-bits source message. In this work, we use BPSK modulation. BPSK allows transmission of one bit per channel that means a time slot is what a transmitter takes to transmit 1 bit. Therefore, the transmission efficiency is given by $\zeta = k/n$ where the number of required time slots are normalized by the bit rate. Since the data rates of source-relay link are greater than the data rate of source-destination link, relay nodes can decode the source information faster than the destination. The source information is decoded at relay r_j once the accumulated information satisfies

$$n_j^{sr} C_{sr_j} \geq k, \tag{2.16}$$

where n_j^{sr} is the number of observed channels by r_j to decode source information correctly. Since BPSK is used as modulation, the data rate, C_{ab} of a link, a-b can be calculated by (Zhang & Zhang, 2013):

$$C_{ab}^{BPSK} = 1 - \frac{1}{2\sqrt{2\pi\gamma_{ab}}} \int_{-\infty}^{\infty} log_2(1+e^{-x}) e^{-\frac{x-2\gamma_{ab}}{2\gamma_{ab}}} dx, \tag{2.17}$$

where $\gamma_{ab} = |H_{ab}|^2 P_a/\sigma_b^2$. In both protocols, as soon as the relay decodes source information, it encodes the receive information using fountain codes and transmit through the pre-assigned orthogonal channel to the destination. In source-feedback protocol, the source node continues its transmission until the receiver successfully decodes the source information. Using BPSK as modulation scheme, in this protocol the destination can correctly decode the source information when the accumulated information satisfies

$$nC_{sd}^{BPSK} + \sum_{j=1}^{R}(n - n_j^{sr})C_{r_jd}^{BPSK} \geq k, \tag{2.18}$$

where, R is the number of participating relay nodes, C_{sd} is the data rate of s-d link and C_{r_jd} is the data rate of r_i-d link. Restricting the modulation scheme to BPSK, the maximum achievable transmission efficiency ζ^* of this protocol can be defined as

$$C_{sd}^{BPSK} + \sum_{j=1}^{R}(1 - \frac{n_j^{sr}}{n})C_{r_jd}^{BPSK} = \zeta^*. \tag{2.19}$$

In non-source-feedback protocol, the source node ceases its transmission after all relay nodes successfully decode source information. Relay nodes transmit towards the destination until the source information is successfully decoded at the destination. Therefore, in non-source-feedback protocol, the destination may reliably decode the source information when the accumulated information satisfies

$$max_n^{sr} C_{sd}^{BPSK} + \sum_{j=1}^{R}(n - n_j^{sr})C_{r_jd}^{BPSK} \geq k, \tag{2.20}$$

where $max_n^{sr} = max\{n_1^{sr}, n_2^{sr}, \ldots n_R^{sr}\}$ is the required time for the relay that has minimum source to relay data rate, $min\ C^{sr} = min\{C_{sr_1}, C_{sr_2}, \ldots C_{sr_R}\}$. The maximum transmission efficiency, ζ^*

of this protocol can be defined as

$$\frac{max\ n^{sr}}{n}C_{sd}^{BPSK}+\sum_{j=1}^{R}(1-\frac{n_j^{sr}}{n})C_{r_jd}^{BPSK}=\zeta^*. \qquad (2.21)$$

2.6 Simulation Results

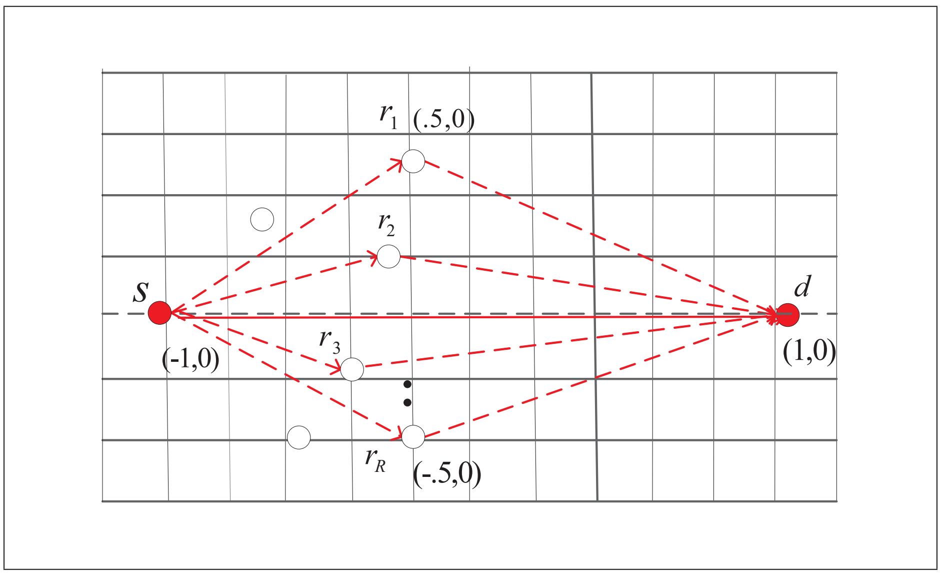


Figure 2.9 Simulation model

The simulation scenario in our study consists of a pair of source CH and destination CH with a number of relay candidates which are placed randomly within the area $-1 \leq x \leq 1$, $-.5 \leq y \leq .5$, where the source is placed at (-1,0) and the destination is at (1,0) as shown in Figure 2.9. For simplicity, the Gaussian noise at each node is assumed to be of the same variance σ_0^2, and each node has similar transmit power P_0. Without loss of generality, we assume a unit path loss between source and destination, i.e., $G_{sd}=1$. We then have $G_{sr_i}=\left(\frac{d_{sd}}{d_{sr_i}}\right)^{\alpha}$ and $G_{r_id}=\left(\frac{d_{sd}}{d_{r_id}}\right)^{\alpha}$ where $\alpha=2$ is the path loss coefficient and n 898d_{sd}, d_{sr_i} and d_{r_id} are the distances between source and destination, source and i^{th} relay node r_i, and destination and r_i, respectively. We assume free-space path loss, so we have $\alpha=2$.

2.6.1 Performance of Relay Selection Algorithm

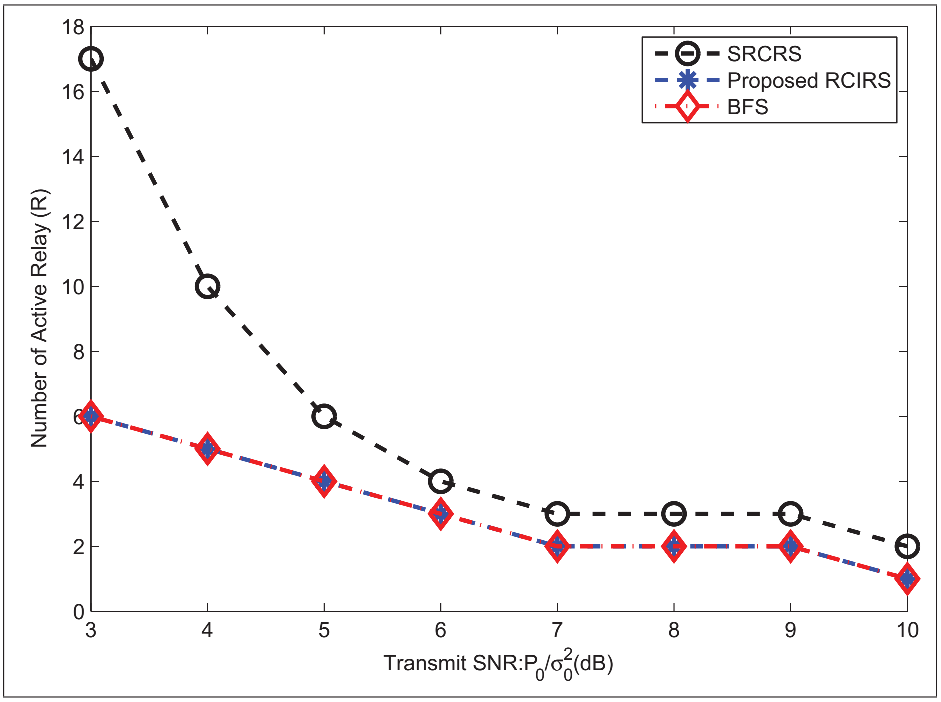


Figure 2.10 Number of active relays vs transmit SNR(dB) $(N = 25, C_{target} = 4$ bits/sec)

In this subsection, we conduct simulations to evaluate the performance of the proposed relay selection algorithm. The channels between all nodes are assumed to be frequency-flat block-fading Rayleigh channels. We compare our relay selection method with source relay channel based relay selection (SRCRS) as in (Molisch *et al.*, 2007) and (Bali *et al.*, 2010) where relay nodes are selected based on the channel quality between source and relays without considering the relays and destination links.

Figure 2.10 shows the number of active relays as a function of transmit SNR, P_0/σ_0^2 for three schemes including RCIRS, SRCRS and Brute-Force search (BFS) having a data rate constraint, $C_{target} = 4$ $bits/sec$. We assume $N = 25$. Nodes that participate in the cooperative group until the destination successfully decodes the source message are defined as active relays. From Figure 2.10 it is observed that SRCRS employs more relay nodes than RCIRS to meet the target data rate. This is because in SRCRS, node is included in the cooperative group as soon as it decodes the source information successfully with the possibility of having poor link towards destination. It is also observed that RCIRS employs equal number of relays as BFS to meet the data rate requirements with less computation complexity. Moreover, it is shown that at lower transmit SNR region, a large number of relays are required to meet the data rate requirements, which decreases gradually at higher SNR region.

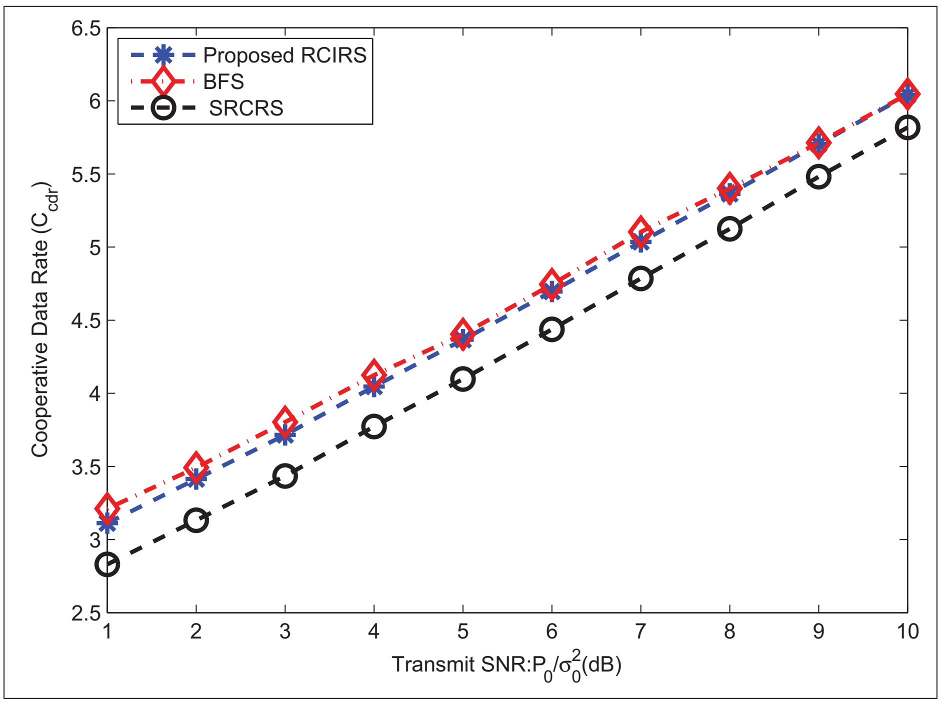


Figure 2.11 Cooperative data rate vs transmit SNR(dB) $(N = 25, R = 4)$

Figure 2.11 illustrates the cooperative data rate vs transmit SNR, P_0/σ_0^2 for different schemes with parameters, $R = 4$ and $N = 25$. It is observed that RCIRS almost attains the performance of BFS as expected and outperforms the SRCRS. This is because in RCIRS, node is added in the cooperative group based on the data rate of source to relay and relay to destination links while in SRCRS, only the data rate of source to relay is considered.

Table 2.2 Comparison among different relay selection methods $(N = 25)$

$P_0/\sigma_0^2(dB)$	Method	C_{cdr} (bits/sec)	R
4	BFS	4.1996	8
	RCIRS	4.1812	8
	SRCRS	4.1996	24
8	BFS	5.4882	7
	RCIRS	5.4799	7
	SRCRS	5.4882	20

Table 2.2 lists the maximum achievable cooperative data rate, C_{cdr} and corresponding number of active relay nodes, R for different value of P_0/σ_0^2 and number of initial relay candidates, $N = 25$. It is noted that SRCRS achieves the same C_{cdr} as BFS by employing more number of relay nodes than that of the BFS. BFS maximizes C_{cdr} by employing minimum number of relay nodes with the cost of huge computation complexity (e.g., $2^{25} - 1$). It is worth noting that our proposed RCIRS almost attain the C_{cdr} of BFS by employing the same number of relay nodes as of BFS with less computational complexity. At $P_0/\sigma_0^2 = 8$ dB, the achievable cooperative data rate of RCIRS is 5.4799 bits/sec, which is only 0.15% less than that of BFS.

2.6.2 Performance of Fountain Coded Cooperative Protocols

In this subsection, we compare the performance of the proposed fountain coded cooperative protocols under different relay selection methods with that of noncooperative direct communication scheme. Figure 2.12 presents the performance of the noncooperative direct transmission (DT) and proposed protocols in terms of transmission efficiency, ζ as a function of transmit SNR, P_0/σ_0^2 under different relay selection methods for parameter $R = 2$. It is observed that cooperative protocols dramatically improve the transmission efficiency than the direct transmission between two neighboring CHs. We observe that source-feedback based protocol (SF-protocol) always outperforms the non-source-feedback based protocol (NSF-protocol) regardless of the relay selection method. The reason of the better performance of SF-protocol, is that in this protocol relays and source continue their transmission until the destination sends

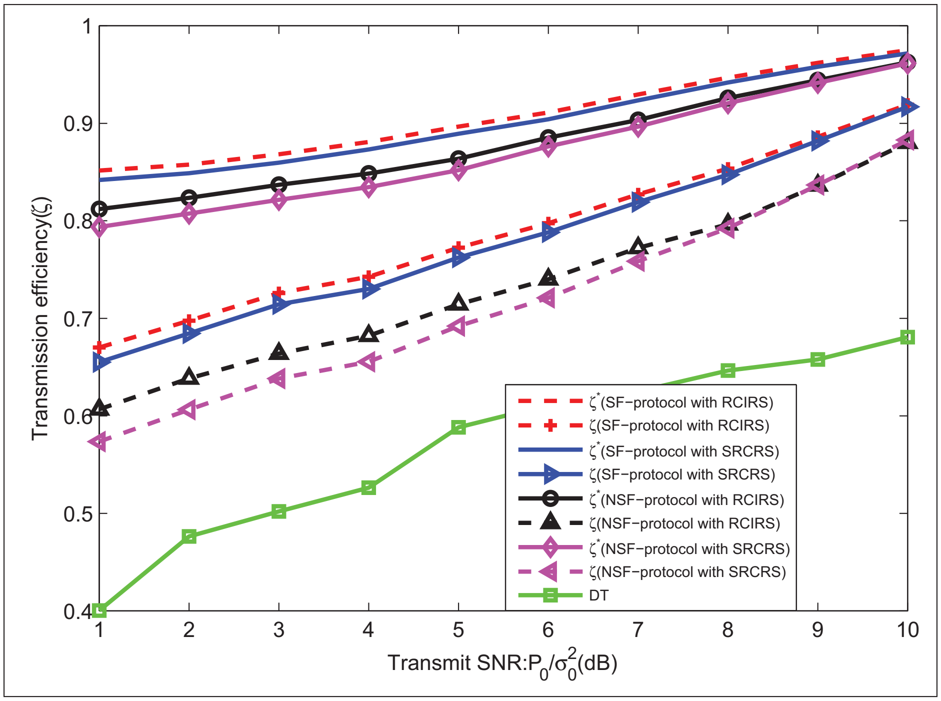


Figure 2.12 Transmission efficiency vs transmit SNR in Rayleigh channel

back an ACK, whereas in the NSF-protocol, the source stops its transmission after the relay nodes decode the message successfully. It is also observed that the performance of the protocols are influenced by the relay selection methods. Any protocol with RCIRS outperforms its SRCRS counterpart. However the impact of relay selection method is more noticeable at NSF-protocols than the SF-protocols. It is observed that SF-protocol with RCIRS improves the transmission efficiency 2.22% more than that of SF-protocol with SRCRS while NSF-protocol with RCIRS improves the transmission efficiency 5.45% more than that of NSF-protocol with SRCRS. This is because in the SRCRS method, relay nodes are selected based on the source-relay link without considering the relay-destination link. Therefore, the possibility of having poor relay-destination link of the selected relay nodes and the suspension of the source transmission in NSF-protocol, significantly degrade the performance of the NSF-protocol with SRCRS.

The maximum achievable transmission efficiency, ζ^* of both protocols under different relay selection methods are also presented in Figure 2.12. It is observed as transmit SNR increases, the ζ^* curve reaches the asymptote of 1, due to the BPSK modulation scheme. Comparing the achieved transmission efficiency, ζ with the maximum achievable transmission efficiency, ζ^* for either protocol, we notice that there is an obvious rate loss, which is due to the suboptimality of the Raptor codes. As shown in the figure, at $P_0/\sigma_0^2 = 10 \ \ dB$ under RCIRS method the rate loss of SF-protocol and NSF-protocol are 5.7% and 9%, respectively.

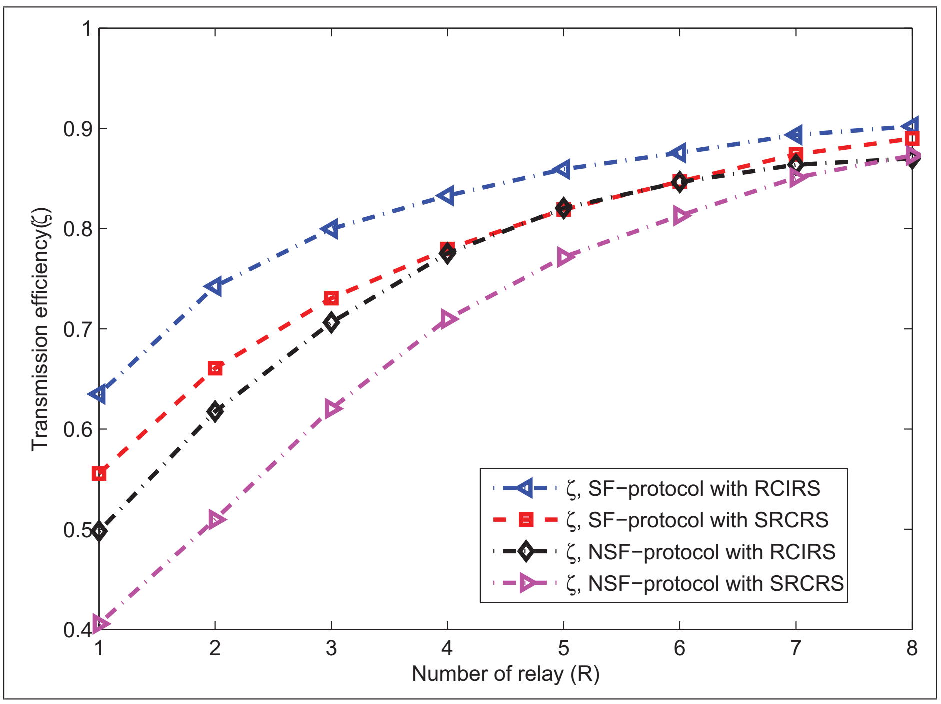


Figure 2.13 Transmission efficiency vs number of relay in Rayleigh channel

In Figure 2.13 we illustrate the transmission efficiency of the proposed protocols as a function of R for fixed transmit SNR, $P_0/\sigma_0^2 = 6 \ \ dB$. We observe that protocols with RCIRS method outperforms the protocols with SRCRS method. We also observe that the performance of the protocols always improves with the involvement of more relays. However, at fixed SNR, the performance difference of proposed protocols under different relay selection methods gradually decrease with the increase of relay nodes as shown in Figure 2.13. This is because, more relays render more mutual information to destination and reduces the impact of direct link at destination.

2.7 Conclusion

Reliable data transmission is an important condition of many applications in IoT paradigm. With respect to this fact, in this chapter we focus on reliable inter-cluster communication and develop fountain coded cooperative schemes to facilitate the data communication between two neighboring CHs. Particularly, we first propose an rateless coded increment relay selection (RCIRS) algorithm to judiciously select relay nodes so that the required data rate is achieved with the minimum number of participating relays. Our proposed relay selection algorithm best suits for adaptive systems in which the data rate between two clusters can be adjusted to attain the required end to end data rate to ensure quality of services of the systems. It shows that our proposed RCIRS employs less relay nodes than source relay channel based relay selection (SRCRS) and nearly attains the cooperative data rate obtained by brute force search (BFS). We then develop source-feedback and non-source-feedback based protocols to improve the data transmission between two neighboring CHs. Regardless of the protocol used, transmission efficiency can always be improved by involving more relay nodes for cooperation at the cost of additional complexity. Simulation results demonstrate that our design can significantly improve the end-to-end performance between two CHs than that of non cooperative direct transmission. It is also observed that the source-feedback based protocol always outperforms the non-source-feedback one in terms of transmission efficiency regardless of relay selection method.

CHAPTER 3

EFFICIENT COMMUNICATIONS BY COUPLING NETWORK AND FOUNTAIN CODING

3.1 Introduction

Cooperative communications is a potential candidate to combat the effects of channel and exploit the spatial diversity gains without having multiple antennas at each node. This is achieved by utilizing the advantages of wireless broadcast channel and allowing the neighbor nodes to relay the overheard information to the destination. Destination thus receives multiple replicas of the signals from independent fading paths and achieves diversity even though it is equipped with a single antenna (Laneman et al., 2004). However, the resulting achievement in diversity comes at the cost of a loss of spectral efficiency. In conventional relaying, one relay is restricted to serve only a single source at a particular time. Therefore, in large networks this conventional relaying becomes increasingly bandwidth inefficient due to the allocation of orthogonal channels to forward the signals of different sources.

One way to overcome this bandwidth bottleneck is to use wireless network coding (Ahlswede *et al.*, 2000), in which instead of simply forwarding a packet, a relay node may encode several incoming packets into one or multiple outgoing packets and thus, reducing the number of transmissions. Network coding was first introduced by (Ahlswede *et al.*, 2000) to enhance the multicast capacity in errorless wired network. However, the broadcast nature of wireless networks and the diversity of the links make network coding more attractive in wireless networks. For reliable communication network coding allows the intermediate nodes to generate redundant network-coded packets in a distributed manner thus provides the redundancy across the packet level. Like erasure correcting coding, these redundant packets can be used in upper layer for error recovery. Redundancy can also be added inside each packet, known as error-correction coding. Channel coding is a error-correction technique that is implemented at the physical layer to recover erroneous bits/symbols through redundant parity-check bits/symbols appended to a packet.

However, in the conventional method (Berger *et al.*, 2008) erasure correction coding and channel coding are treated separately. Therefore, erasure correction decoding cannot take advantage of the redundant information in the packets that fail channel decoding. Similarly at physical layer, channel decoding of each packet cannot be benefited by the redundancy across packet levels. In recent years, a number of research efforts have been found on unifying two types of coding schemes (Wilson *et al.*, 2010; Liang *et al.*, 2011; Guo *et al.*, 2012), that shows great potential in improving link robustness and system capacity. However, previous works(Wilson *et al.*, 2010; Liang *et al.*, 2011; Guo *et al.*, 2012) addressed the fixed rate code. Many fixed rate codes have sound performance and can approach the channel capacity of non-fading channels, but they cannot fully adapt to channel conditions and require flexibility in aligning with the network code. Since the code rate is fixed, decoding failures may occur when the channel degradation exceeds the error-correction capability of the codes. Hence, the performance of such coding over heavily impaired channel may not satisfactory due to the overhead of feedback and end to end delay.

In contrast to typical fixed rate code, fountain code is a rateless code where the source unconscious of channel state information (CSI) can generate as many encoding symbols as needed by simply performing modulo-2 operation among the source symbols. In the previous works (Molisch *et al.*, 2007; Liu & Lim, 2009; Castura & Mao, 2007; Nessa *et al.*, 2012), it has been incorporated into cooperative relay systems as a desirable solution for data communication. In a typical cooperative relay system, relays assist multiple source to forward their information to the destination. Such a scenario can be modeled as M-N-1 system, which consists of M sources (eg. M source CHs), N relays and coordinator node, D. In this chapter, we develop joint network and fountain coding (JNFC) schemes for a M-N-1 system where $M = N = 2$. The idea behind this is to couple network and channel coding techniques simultaneously in the physical layer so that the redundancy provided by network coding can be used to support the channel coding for better error protection (Nessa *et al.*, 2014a,b). We apply different coding techniques to generate network codes at relay nodes. We compare the performance of proposed schemes and separated network and fountain coding (SNFC) where the redundancy provided by network coding is only useful if channel coding is succeeded. As a result, the packets that fail the channel decoding are wasted.

The rest of the chapter is organized as follows. In Section 3.2 we first review the existing effort in literature on unifying channel code and network coding. Then in Section 3.3 we describe some preliminaries for the proposed JNFC scheme. We then present the proposed scheme using a simple topology with two sources and two relays in Section 3.4. Encoding and decoding procedures of proposed JNFC is presented in Section 3.5. Section 3.6 presents simulation results, followed by Section 3.7 to conclude this chapter.

3.2 Related Work

Channel coding is indispensable part in communication networks to provide reliable communication. It has been widely employed in practical wired and wireless systems. The history of channel coding began with Claude Shannon's information theory (Shannon, 1948). Different channel coding techniques such as Reed-Solomon (RS), convolution, Turbo and low-density parity-check (LDPC) codes are described in (Blake, 2005). Many of them approach the Shannon capacity over non-fading channels and have sound performance. However, when a channel experiences slow and deep fading, the performance of channel coding degrades dramatically. In such a case, the communication through the channel cannot continue and packet loss will occur. To provide more reliable communication, coding is also applied in the network layer to generate redundant packets for extra error protection. Therefore separate network channel coding is naturally formed. In (Nguyen *et al.*, 2009), authors combined network coding (NC) with traditional FEC to increase the bandwidth efficiency for one-source, multisink one-hop networks. In this paper after receiving a NACK from a receiver, source does not immediately retransmit the requesting packet rather it maintains a queue of lost packets and combined different lost packets in such a way to allow the receivers to recover their lost packets simultaneously with a single transmission from it. In (Tran *et al.*, 2009), authors utilize network coding to implement Type-I Hybrid-ARQ for one-source, multisink one-hop networks. It is actually an extension of the previous work (Nguyen *et al.*, 2009) with a conjunction of appropriate channel coding for different channel conditions. In (Berger *et al.*, 2008), authors theoretically analyze the optimization problem in joint erasure-correction and error-correction coding schemes. All these studies treat two levels of codes separately and do not fully exploit the precious redundant information.

There are also some work that can be categorized by distributed channel coding where the procedure of channel coding is distributed to different nodes in the network. For example, in (Bao & Li, 2006a), authors proposed adaptive network coded cooperation (ANCC) for multiple transmitters sending data to a common receiver where low-density generator matrix (LDGM) codes are dynamically constructed. Later, in (Bao & Li, 2006a), they analyze the outage properties of ANCC. In general, ANCC is not practical as it is difficult to have such a large number of nodes in real networks. Recently, many active studies jointly design network-channel codes to fully exploit the redundancy in both channel and network codes. Following this stream, Bao and Li extended their previous work (Bao & Li, 2006b, 2008) to generalized adaptive network coded cooperation (GANCC) (Bao & Li, 2011) where channel coding is integrated with network coding. In (Hausl *et al.*, 2005), authors presented iterative network and channel decoding on a Tanner graph. They also proposed joint network and channel coding framework for multiple access relay channel (MARC) based on low-density parity-check (LDPC) code in (Hausl & Dupraz, 2006) and for two way relay networks in (Hausl & Hagenauer, 2006). Later on (Winkelbauer & Matz, 2012) proposed joint network-channel coding schemes for multi-way relay networks. In (Kang *et al.*, 2008) and (Yang & Koetter, 2007), authors proposed iterative network and channel decoding considering the situation when the relays cannot perfectly recover packets. However, the aforementioned joint network-channel coding schemes were designed for small-scale wireless networks with specific topologies. In (Guo *et al.*, 2012) the authors developed non binary joint network-channel decoding (NB-JNCD) for large networks. It has been shown that NB-JNCD outperforms binary LDPC JNCD. However, the aforementioned works addressed the fixed rate codes that provide a stable error performance when the channel state information is available at the transmitter. In this case the transmitter estimates the loss rate and choose the closest code rate to adapt the channel conditions. Therefore, decoding failures may occur when the channel can not estimate correctly or channel be changed during the transmission. If that occur, a negative acknowledgement (NACK) is sent to sender after every detection of corrupted message at receiver that increases the end to end delay over heavily impaired channel. In the multihop communications, the end-to-end delay may exceeds the allowable delay budget.

In recent years researchers have also paid attention on coupling fountain and network coding. For example, in (Puducheri *et al.*, 2007), authors proposed a low complexity combining operation at relays for a multiple access relay system where messages from M sources are encoded by a rateless code that performs like a LT code in data recovery and completes the network coding inherently. In this work, source nodes generate their information using deconvolved soliton distribution (DSD) as in degree distribution and transmit to relay. Relay node then constructs LT codes by merely XOR-ing the received symbols from different sources and transmits to the destination. The performance is evaluated in binary erasure channel (BEC) without considering the direct link between source and destination. Later on in (Gong *et al.*, 2010) for a multiple access relay system, raptor code based two combining schemes, namely, raptor coding (RC) and superposition coding (SC) at the relay have been proposed that improve the performance gain significantly. In Zhang & Zhang (2013), authors proposed a joint network-channel coding (JNCC) with Raptor code for a 2-1-1 system. They optimized the degree distribution of the Raptor code and evaluated their proposed JNCC in terms of BER and throughput performance. However, the aforementioned joint network-channel coding schemes were designed for small-scale wireless networks with specific topologies. Moreover, they require the transmissions of each node to be well scheduled.

3.3 Evolution from SNFC TO JNFC

In this section, we first present the background of fountain coding (in particular LT coding), random linear network coding and network coding with robust soliton distribution also known as modified LT code(MLT) (Puducheri *et al.*, 2007). We then introduce the separated network and fountain coding (SNFC) system and that will be used as the reference system to compare the performance of JNFC.

3.3.1 LT: Encoding and Decoding

The concept of fountain codes was first presented by Byers et al.(Byers *et al.*, 1998/10/). The most interesting benefit of fountain code is that transmission reliability can be assured without requiring channel state information. Although fountain codes were originally designed for erasure channels, a lot of effort has been dedicated to their extension to general discrete memoryless channels, additive white Gaussian noise (AWGN) channels, and fading channels. A universal fountain codes should have two properties: First, infinite encoded data can be generated from finite source data. Second, the receiver should be able to reconstruct the source data from any set of the encoded data with an efficient decoding process. However, not all fountain codes in use can meet the above two properties.

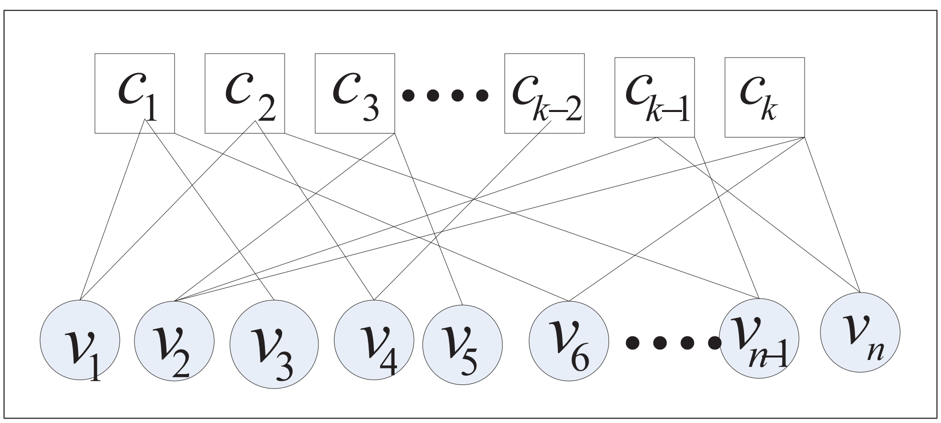


Figure 3.1 A factor graph representation of LT code symbols: $v_1 = c_1 \oplus c_2$, v_1 has degree 2 and v_2 has degree 3

LT codes (Luby, 2002) are the first realization of a true digital fountain codes. In LT codes, the symbol length for the code can be arbitrary, from one-bit binary symbols to general k-bit symbols where k is the length of the information. Each LT code symbol is generated by the following encoding process:

- First a degree d is chosen for an encoding symbol. The degree is chosen randomly from a given degree distribution $\rho(d)$;
- Choose d distinct information symbols uniformly at random. They will be neighbors of the encoding symbol;
- Then chosen original symbols are XOR-ed to create an encoded symbol.

The canonical representation of fountain codes are factor graphs (Kschischang *et al.*, 2001). A factor graph is a bipartite graph where nodes in the first set represent original symbols referred as input symbols and the nodes in the second set are the encoded symbols referred as output symbols. The input symbol c is a neighbor of v if there is an edge between them. The degree of an output symbol is the number of edges originated from that particular node. A factor graph representation of encoded symbols is shown in Figure 3.1 that is truncated to length n. The decoder can recover information symbols with the following three-step process, which is called LT process:

- At first step, the decoder identifies all output symbols of degree one (i.e., those connected to a single input symbol) in the Tanner graph. If there exist no such symbols, the decoding process terminates;
- The input symbols connected to output symbols of degree one are directly decoded and the edges between them is deleted;
- Finally, each the decoded input symbol c is XORed with the every output symbol v to which c is connected and the edge between c nd v is deleted.

The decoding process continues iteratively by following the above three steps. The decoding process succeeds if all information symbols are covered by the end.

3.3.2 Degree Distribution of LT Codes

The probability distribution on the random degree of encoding symbols, $\rho(d)$ is the critical part of the LT codes design to ensure complete recovery of the original data from the minimum number of encoding symbols. In fact, the encoding/decoding complexity and error performance are regulated by the degree distribution of LT code. For better performance, the degree distribution should be such that a small number of encoding symbols must possess high degree, so that all input symbols get connected with output symbols and a large number of output symbols must have low degree, so that the decoding process can get started, and keep going. The optimal distribution of degrees for constructing LT codes is the Robust Soliton Distribution (RSD), proposed by Luby is given bellow (Luby, 2002):

$$\rho(i) = \begin{cases} \frac{\mu(i)+\vartheta(i)}{\beta}, & \text{for} \quad 1 \leq i \leq k \end{cases}, \tag{3.1}$$

where

$$\beta = \sum_{i=1}^{k} (\mu(i) + \vartheta(i)). \tag{3.2}$$

Here $\mu(i)$, the Ideal Soliton distribution and $\vartheta(i)$ are given by

$$\mu(i) = \begin{cases} \frac{1}{k}, & \text{for} \quad i = 1, \\ \frac{1}{i(i-1)}, & \text{for} \quad 2 \leq i \leq k. \end{cases} \tag{3.3}$$

$$\vartheta(i) = \begin{cases} \frac{R}{k}, & \text{for} \quad 1 \leq i \leq \frac{k}{R} - 1, \\ \frac{R}{k} ln \frac{R}{\delta}, & \text{for} \quad i = \frac{k}{R}, \\ 0, & \text{otherwise}, \end{cases} \tag{3.4}$$

where δ is the allowable failure probability and the parameter R represents the average number of degree one encode symbol and is defined as

$$R = \lambda \; ln(k/\delta)\sqrt{k}. \tag{3.5}$$

It has been shown that for a suitable chosen λ the decoder can recover the data from $n = k\beta = k + \lambda.\sqrt{k}.ln^2(k\delta)$ encoded symbols with probability at least $1 - \delta$ (Luby, 2002). It is observed that RSD is composed of more than 50% of encoded packets of degree 1 or 2 allowing to bootstrap belief propagation, and an average degree of $\mathscr{O}(\log k)$ resulting in low complexity decoding.

3.3.3 Random Linear Network Coding (RLNC)

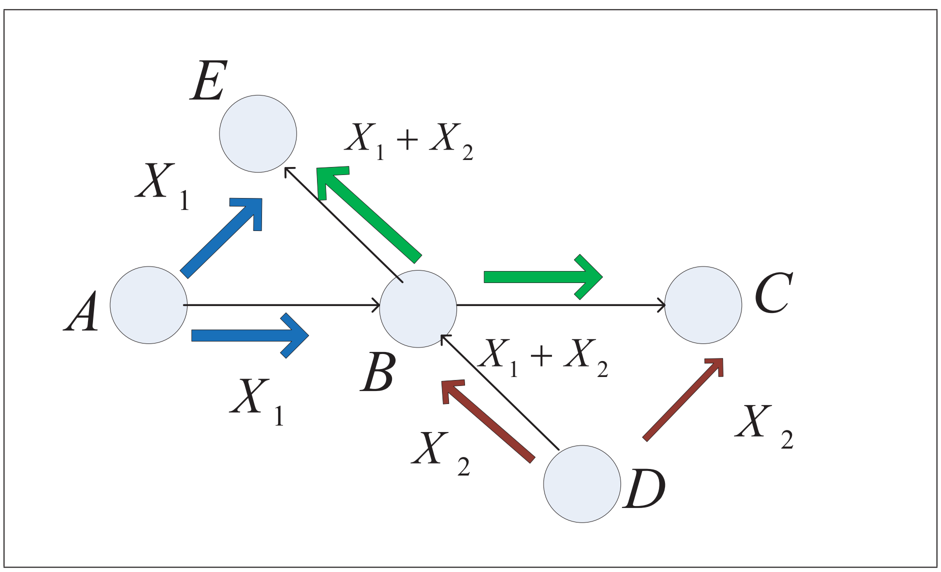


Figure 3.2 Example of network coding

Work on network coding was first started by (Ahlswede *et al.*, 2000) to enhance the multicast capacity in errorless wired network. However, the broadcast nature of wireless networks and the diversity of the links make network coding more attractive in wireless networks. While in traditional cooperative communications intermediate nodes just forward the received packets, in network coding intermediate nodes process the data packets before forwarding. The basic concept of the network coding can be given by an example illustrated in Figure 3.2 where, A wants to send a packet, X_1 to C that is out of reach of A. To send its packet to C, A takes help of B. B receives packet form A and forward to C. Similarly D wants to send its packet X_2 to E by the help of relay B. B receives from both A and D and transmits the network code X_1+X_2 to C and E simultaneously. E can then extract X_2 from received X_1+X_2 and C can extract X_1 from received X_1+X_2 by accomplishing XOR operation.

Though the idea of network network coding was first proposed for muticasting wired network, the broadcast nature and other inherent characteristics of the wireless medium make it advantageous to use in data communication in wireless networks. For information multicast over multihop wireless networks, network coding based solutions offer superior performance compared to state-of-the-art forwarding based solutions. Network coding over Galois fields (GFs) is an efficient approach to increase the throughput of multi-source cooperative diversity systems (Chen *et al.*, 2006; Yu *et al.*, 2007). Several other network coding schemes have been proposed for general multisource cooperative networks, such as physical layer network coding (PNC) for two way relaying (Zhang *et al.*, 2006) and complex field network coding (CFNC)(Wang & Giannakis, 2008). Even for unicast (one-to-one) communications, network coding offers an attractive solution because its robustness can better deal with the dynamics in mobile networks (e.g., vehicular networks, adhoc networks formed in the battlefield). Through analysis and simulation, it has been shown in the previous studies (Ho *et al.*, 2006; Guo *et al.*, 2009; Koetter & Medard, 2003) that network coding offers benefits in diverse dimensions of communication networks such as throughput, energy efficiency, reliability and resource utilization, security and resilience to link failures.

In our proposed joint network and fountain coding we choose random linear network coding because the randomness property of this code makes it more suitable to apply in large large network where each intermediate node can perform distributed operation on the received packets without interrupting others. Let relay r receives M incoming packets $\{x_1, x_2, \ldots, x_M\}$, relay then linearly combines these M packets to compute $M'(M' \geq M)$ outgoing packets, denoted as $\{y_1, y_2, ..., y'_M\}$ where $y_i = \sum_{j=1}^{M} g_{ij}x_j$. The coefficient gij is picked randomly from GF(2). The set of coefficients $\{g_{i1}, ..., g_{iM}\})$ is referred as the encoding vector for y_i and is carried in packet as overhead.

3.3.4 Network Coding with Robust Soliton Distribution

One of the attractive features of LT codes is low complexity decoding which is accomplished using Belief Propagation (BP) algorithm that recovers source information k on average $\mathscr{O}(k.\ ln(k))$ symbol operations. However, the efficiency of BP depends on the statistical properties of encoded symbols degree distribution. Therefore, the degree distribution of the network coded symbols must match RSD to get better performance. More specially, the network node should generate the network coded symbols in such a way so that the structure of LT codes is preserved.

The construction of LT codes from two or more fountain codes is proposed in (Puducheri *et al.*, 2007) where each source node encodes its data set onto an LT-like codewords according to a degree distribution $p(.)$ and sequentially transmit to relay node. Relay node then generates new code symbol Y by selectively XOR-ing each pair of symbols it received from S_1 and S_2 and transmits to the destination. The result in sequence of symbols that is referred as a modified LT (MLT) codes follows RSD in degree and has erasure correcting properties similar to those of an LT codes. To determine $p(.)$, the author employed deconvolution of the RSD. In this aim, $\rho(.)$ is splitted into two distributions $\rho_1(.)$ and $\rho_2(.)$ where

$$\rho_1(i) = \begin{cases} 0, & \text{for } i = 1, \\ \frac{\mu(i)+\vartheta(i)}{\beta_1}, & \text{for } \ 2 \leq i \leq \frac{k}{R} - 1, \\ \frac{\mu(i)}{\beta_1}, & \text{for } \ \frac{k}{R} \leq i \leq k, \end{cases}$$

with the normalization factor β_1 given by

$$\beta_1 = \sum_{i=2}^{k} \mu(i) + \sum_{i=2}^{k/R-1} \vartheta(i), \tag{3.6}$$

and

$$\rho_2(i) = \begin{cases} \frac{\mu(1)+\vartheta(1)}{\beta_2}, & \text{for} \quad i = 1, \\ \frac{\vartheta(\frac{k}{R})}{\beta_2}, & \text{for} \quad i = \frac{k}{R}, \\ 0, & \text{otherwise}, \end{cases} \tag{3.7}$$

with the normalization factor β_2 given by

$$\beta_2 = \mu(1) + \vartheta(1) + \vartheta(\frac{k}{R}). \tag{3.8}$$

Finally, the Deconvolved Soliton Distribution (DSD), $p(.)$ is given by (Puducheri *et al.*, 2007)

$$p(i) = \gamma . f(i) + (1-\gamma).\rho_2(i), \quad \text{for} \quad 1 \leq i \leq k/2, \tag{3.9}$$

with the parameter γ

$$\gamma = \sqrt{\frac{\beta_1}{\beta}}, \tag{3.10}$$

and

$$f(i) = \begin{cases} \sqrt{\rho_1(2)}, & \text{for} \quad i = 1, \\ \frac{\rho_1(i+1) - \sum_{j=2}^{i-1} f(j) f(i+1-j)}{,} & \text{for} \quad 2 \leq \ i \leq \frac{k}{2}, \\ 0, & \text{for} \quad \frac{k}{2} \leq i \leq k. \end{cases} \tag{3.11}$$

It is observed that DSD is dominated by degree 1 while RSD is dominated by degree 2. The degree distribution of RSD and DSD for $\delta = .5$ and $\lambda = .2$ is presented in Table 3.1

Table 3.1 THE RSD AND DSD FOR DEGREES $1 \leq d \leq 5$
$(\lambda = 0.2, \delta = 0.5)$
Taken from Puducheri *et al.* (2007)

Degree	**RSD**	**DSD**
1	.0033	.7028
2	.4915	.1171
3	.1642	.0489
4	.0823	.0271
5	.0495	.0174

Relay receives two symbols X_1 and X_2 from S_1 and S_2 respectively. Then the network coded symbol Y is generated in the following way (Puducheri *et al.*, 2007):

- Let d_1 and d_2 are the degree of X_1 and X_2 respectively;
- The relay generates two independent random variables U_1 and U_2, each uniformly distributed on [0, 1];
- The relay then generates two binary random variables b_1 and b_2 as follows:

$$b_i = \begin{cases} 1, \ if \ (d_i = 1 \ and \ U_i \leq 1 - \frac{\gamma.f(1)}{p(1)}) \ , \\ 1, \ if \ (d_i = k/R \ and \ U_i \leq 1 - \frac{\gamma.f(k/R)}{p(k/R)}); \\ 0, \ otherwise. \end{cases}$$

- After that relay transmits the binary random variable Y defined as follows (Puducheri *et al.*, 2007):

$$Y = \begin{cases} X_1 \oplus X_2, & if\ b_1 = b_2 = 0, \\ X_1, & if\ b_1 = 1\ and\ b_2 = 0, \\ X_2, & if\ b_1 = 0\ and\ b_2 = 1, \\ X_1 or X_2 & if\ b_1 = 1\ and\ b_2 = 1. \end{cases} \tag{3.12}$$

The degree of each resulting symbol Y is either the sum of the degrees, or the individual degree of one of the received symbols.

3.3.5 Separated Network and Fountain Codes (SNFC)

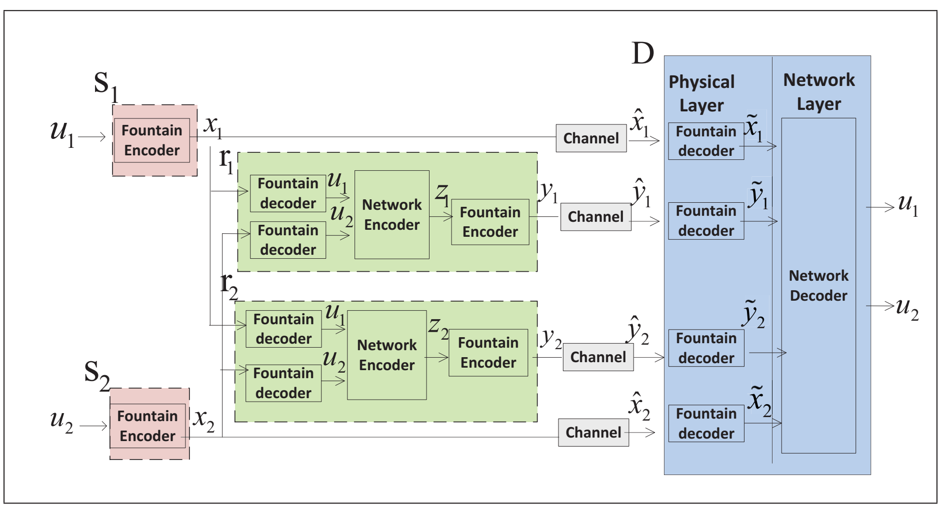


Figure 3.3 Block diagram of separated network and fountain coding

Spatial diversity is one of the ways to combat fading over wireless channels. One way to gain diversity through network coding in noisy channels is to treat network and channel coding separately where channel coding is used in the physical layer for each transmission then on the network layer network coding is performed on the error free packets provided by the lower layers.

Figure 3.3 presents the block diagram of SNFC system where source nodes, S_1 and S_2 encode their data packets u_1 and u_2 respectively using fountain codes and transmit to destination. Relay nodes keep accumulating information from source to destination transmission. As soon as relay r_i decodes sources information correctly, it performs network coding on the original packets u_1 and u_2 at network layer. The network encoder is a modulo-2 addition. Then the output of network encoder z_i, is encoded using fountain code and transmitted to destination. The destination collects information from four channels and starts to decode when the received information from each channel is slightly greater than the original symbols. Let destination starts decoding on the four set of symbols $\hat{x_1}$,$\hat{x_2}$, $\hat{y_1}$ and $\hat{y_2}$ using fountain decoder. The four decoder make a hard decision and deliver their estimates $\tilde{x_1}$,$\tilde{x_2}$,$\tilde{y_1}$ and $\tilde{y_2}$ to the network layer with an indication to indicate whether its estimate is error-free. If one of the two estimates $\tilde{x_1}$, or $\tilde{x_2}$, not error-free and either $\tilde{y_1}$ or $\tilde{y_2}$ are error-free, the network decoder retrieves the corrupted packet by performing module-2 addition between the error free packet among $\tilde{x_1}$, or $\tilde{x_2}$, and the error free packet among $\tilde{y_1}$, or $\tilde{y_2}$. If both $\tilde{y_1}$, or $\tilde{y_2}$ is error free then any one of them is used to retrieve the corrupted packet. From the above discussion, it is observed in SNFC, the redundancy of network coded packets only utilized when the lower layers deliver error-free packets to the network layer.

3.4 System Model

In this section we consider clustered M2M network where nodes are grouped into number of clusters. Data is aggregated from non cluster head nodes and transferred to respective CHs and then to the destination with the help of the relay nodes. We investigate a system consisting of two source CHs, S_1, and S_2, a destination D and two relays r_1 and r_2 as shown in Figure. 3.4. We assume that all nodes are operated in half-duplex mode, i.e., they can either transmit and receive at a time. To transmit a message u_i of length k bits to a destination D, each source CH S_i generates a large number of code stream using fountain codes. The code stream is then mod-

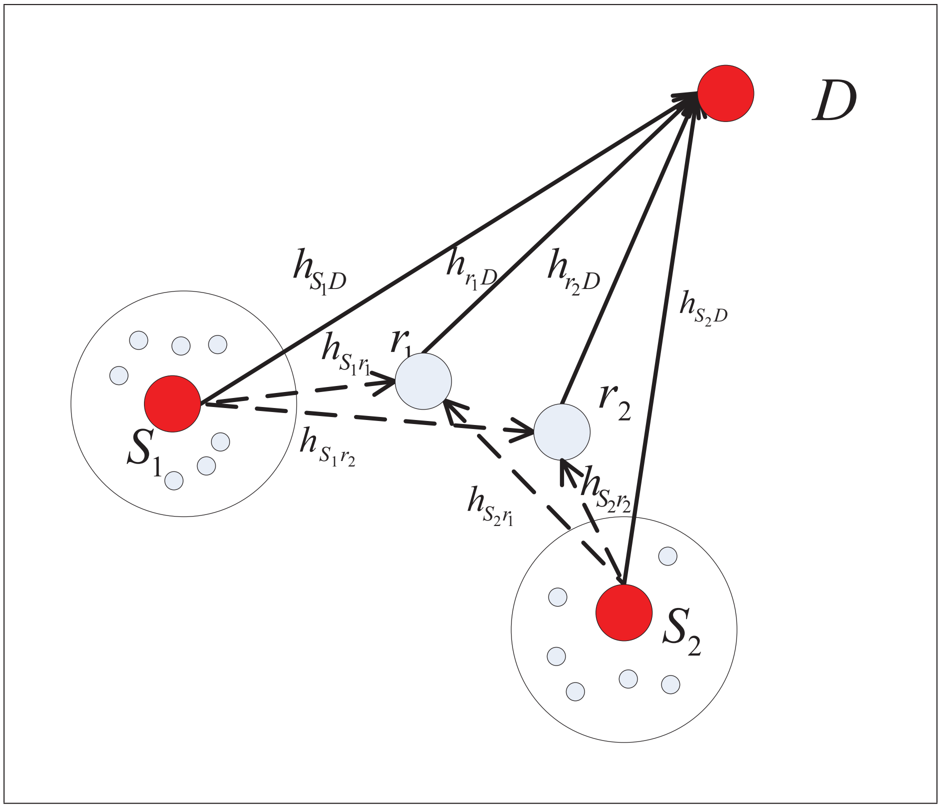


Figure 3.4   Network topology consisting of two sources two relays and sink

ulated and sequentially transmitted to the destination. Relay nodes r_1 and r_2 monitor source transmission and keep accumulating information from source to destination transmission. As soon as the original packets, u_1 and u_2 are successfully decoded at relays, they generate new packets y_1 and y_2 using network coding and fountain coding and send to the destination to provide additional error protection. After passing through wireless channel x_1, x_2, y_1 and y_2 reach at destination as $x_{1'}$, $x_{2'}$, $y_{1'}$, and $y_{2'}$, respectively. A block diagram of the system is depicted in Figure 3.5.

We assume that all the nodes use their own fountain coding routine to encode the data and the encoding degree distribution of the transmitted packets are sent to the destination by an extra robust CDMA channel with long sequence length instead of using identical random-number generator both in encoder and decoder as used in (Liu & Lim, 2009), that needs strict synchronization and increases the complexity in multiple relay network.

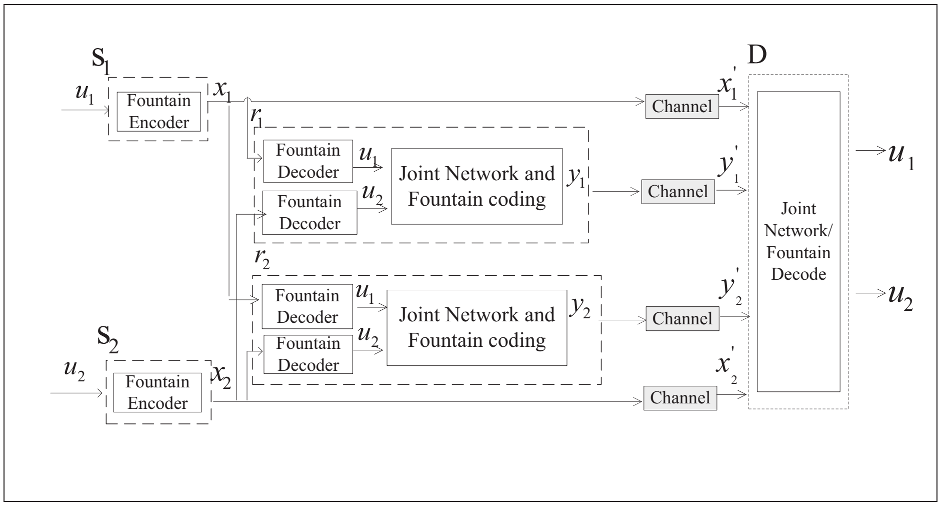


Figure 3.5 Block diagram of joint network and fountain coding system

3.4.1 Channel Model

The propagation channels between different nodes are considered as independent and identically distributed (i.i.d.) Rayleigh fading with additive white Gaussian noise (AWGN). We adopt binary phase shift keying (BPSK) modulation thus the channel model between any transmitter i and receiver j can be written as:

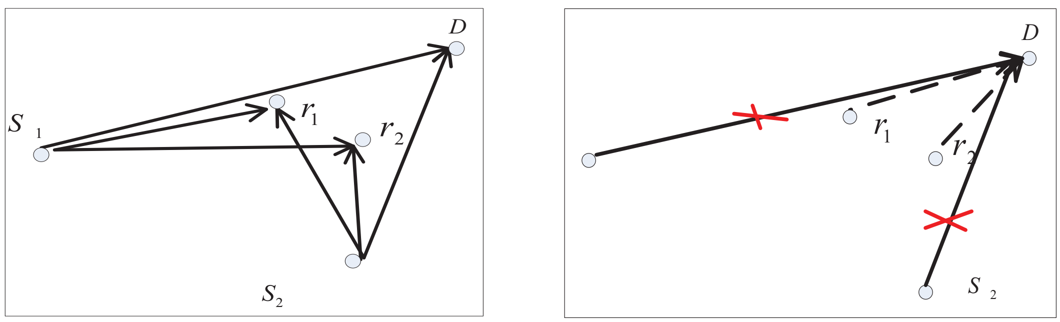
$$y_j[n] = h_{i,j} x_i[n] + n_i[n] \tag{3.13}$$

where $x_i[n]$ is the BPSK modulated transmitted symbol, $y_j[n]$ is the received symbol, $n_i[n]$ is zero mean and additive white Gaussian noise with variance σ_i^2 at time instant $[n]$, $h_{i,j}$ captures the combined effects of symbol asynchronism, fading, quasi-static multipath fading, shadowing, and path-loss between nodes i and j. Therefore, the signal-to-noise ratio (SNR) between node i and j is given by

$$SNR_{i,j} = \frac{P_i |h_{i,j}|^2}{\sigma_i^2} \tag{3.14}$$

where P_i denotes the transmitted power of node i.

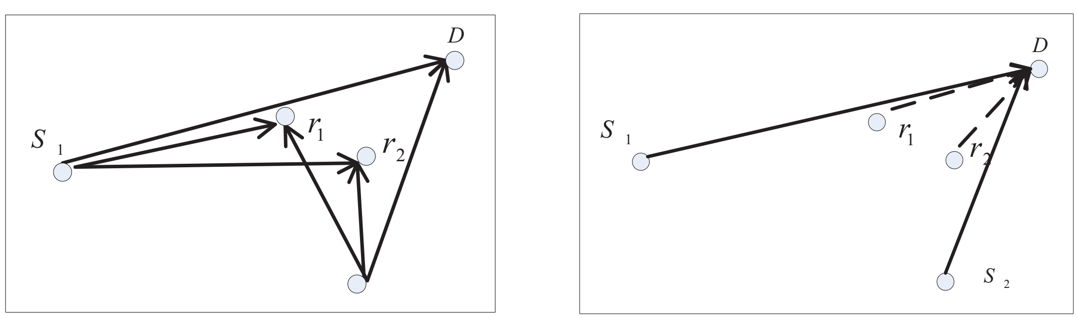
3.4.2 Data Transmission Protocols



a) Phase I: source nodes transmit to relay and destination

b) Phase II: Both relays have decoded information of both sources and start to transmit to destination while both sources stop their transmission

Figure 3.6 Transmission strategy in protocol A



a) Phase I : Source nodes transmit to relay and destination until both relays decode sources information successfully

b) Phase II: Relays have decoded information of both sources nodes and join with source nodes to transmit to destination

Figure 3.7 Transmission strategy in protocol B

Depending on the participation of the source nodes until destination node decodes their information we propose following two protocols:

- Protocol A : In this protocol, in phase I, source nodes generate a large number of symbols and broadcast their data to relays and destination as shown in Figure 3.6 a). After successfully decoding the messages from both sources, relay nodes re-encode the original packets of sources, perform network coding or generate a large LT considering each source information as a subcode and send ACKs to sources. Sources then stop their transmission. In phase II, relay nodes start to transmit to the destination until the destination is able to decode all information as shown in Figure 3.6 b);

- Protocol B: In this protocol, in phase I, source nodes transmit their information to destination using LT codes as shown in Figure 3.7 a). Relays overhear the direct transmission between source nodes and destination. After successfully decoding information of both source nodes, relays generate new coded symbols to transmit to destination. In phase II, both source nodes and relays keep on transmitting until they receive an acknowledgment from the destination indicating that the reception has been successful as shown in Figure 3.7 b).

3.5 JNFC Encoding and Decoding

3.5.1 Code Construction

We assume each source wants to transmit same amount of information to destination D. To transmit a packet u_1 and u_2 of length k symbols to D, S_1 and S_2 generate a large number of encoded symbols, $\{x_1 = x_1(1), x_1(2), ...x_1(n)...\}$ and $\{x_2 = x_2(1), x_2(2), ...x_2(n)...\}$ respectively using LT codes. The set of encoded symbols generated by S_1 is given by

$$x_1 = u_1 \mathscr{G}_n^1, \tag{3.15}$$

where, $\mathscr{G}_n^1$ is the generator matrix of the code symbols that are truncated to length n. Similarly, the set of encoded symbols generated by source S_2 is given by

$$x_2 = u_2 \mathscr{G}_n^2, \tag{3.16}$$

where, $\mathscr{G}_n^2$ is the generator matrix of size $k \times n$. Relay nodes accumulate incoming information from source-destination transmission and attempt to obtain u_1 and u_2. We assume that the relay successfully decoded the source message before the destination. After successfully decoding source symbols, relay nodes re-encode each source information using a suitable degree distribution. Let $\{x_{i1}(1), x_{i1}(2), ...x_{i1}(n)...\}$ and $\{x_{i2}(1), x_{i2}(2), ...x_{i2}(n)...\}$ are generated encoded symbols from u_1 and u_2 respectively at relay r_i. Then r_i performs RLNC on the encoded symbols of two sources or generates a large LT (called MLT) codes considering the information of each source as a subcode. Let y_1 and y_2 are the generated symbols at r_1 and r_2 ,respectively and are represented as

$$y_1 = \alpha_{11} u_1 . \mathscr{G}_n^3 \oplus \alpha_{12} u_2 . \mathscr{G}_n^4, \tag{3.17}$$

$$y_2 = \alpha_{21} u_1 . \mathscr{G}_n^5 \oplus \alpha_{22} u_2 . \mathscr{G}_n^6, \tag{3.18}$$

where the network coding coefficients $\alpha_{ij}(i, j = 1, 2)$ are drawn randomly from GF(2)

3.5.2 Joint Decoding

At the destination four code words $x_1^{'}$, $x_2^{'}$, $y_1^{'}$, and $y_2^{'}$ will be received where each $x_i^{'}$ and $y_i^{'}$ are the part of the codeword x_i and y_i, respectively where $i = 1, 2$. The destination will forms a longer code as follows:

$$[x_1^{'} \; x_2^{'} \; y_1^{'} \; y_2^{'}] = [u_1 \; u_2] \begin{bmatrix} \mathscr{G}_{n'}^1 & 0 & \alpha_{11}\mathscr{G}_{n'}^3 & \alpha_{21}\mathscr{G}_{n'}^5 \\ 0 & \mathscr{G}_{n'}^2 & \alpha_{12}\mathscr{G}_{n'}^4 & \alpha_{22}\mathscr{G}_{n'}^6 \end{bmatrix}. \tag{3.19}$$

where $\mathscr{G}^i_{n'}$ is the part of the generator matrix $\mathscr{G}^i_n$. The length of each generator matrix , $\mathscr{G}^i_{n'}$ and codeword $x^{'}_i$ and $y^{'}_i$ depend on the protocol employed.

In protocol A, each source S_j stops its transmission when both relay nodes decode its message, u_j correctly. We assume $n^{S_j}_{r_i}$ is the number of encoded bits received by relay r_i until successfully decoding the message of S_j where $j = 1,2$. For simplicity we assume that both relays require same number of encoded bits to successfully decode each source message. Therefore all $n^{S_j}_{r_i}, (j = i = 1,2)$ are same and denoted as n'. In this protocol, the transmitted codewords by S_1 and S_2 are given by $x^{'}_1 = u_1\mathscr{G}^1_{n'}$ and $x^{'}_2 = u_1\mathscr{G}^2_{n'}$, respectively where $\mathscr{G}^1_{n'}$ is the $k \times n'$ part of the generator matrix $\mathscr{G}^1_n$ and $\mathscr{G}^2_{n'}$ is the $k \times n'$ part of the generator matrix $\mathscr{G}^2_n$. We assume that the destination receives n_1 number of encodes bits from relays and sources before successfully decoding the source messages. Therefore the remaining $(n_1 - 2n')/2$ bits are transmitted by each relay. In this protocol the codeword transmitted by r_1 is $y^{'}_1 = \alpha_{11}u_1.\mathscr{G}^3_{n'} \oplus \alpha_{12}u_2.\mathscr{G}^4_{n'}$ where the length of generator matrix $\mathscr{G}^i_{n'}, (i = 3,4)$ is $2k \times (n_1 - 2n')/2$. Similarly the number of code bit transmitted by r_2 is $y^{'}_2 = \alpha_{11}u_1.\mathscr{G}^5_{n'} \oplus \alpha_{12}u_2.\mathscr{G}^6_{n'}$ where the length of generator matrix $\mathscr{G}^i_{n'}, (i = 5,6)$ is $2k \times (n_1 - 2n')/2$.

In protocol B , source nodes and relays continues their transmission until the destination decode the source messages correctly. For simplicity like protocol A, we assume that both relay requires same number of encoded bits to successfully decode each source message. We assume the destination receives n_2 number of encoded bits before successfully decoding the source message and the number of received bits from each source in phase 1 is $n^{'}_1$. Therefore, in phase 1, the transmitted codewords by S_1 and S_2 are given by $x_{11'} = u_1\mathscr{G}^1_{n^{'}_1}$ and $x_{21'} = u_1\mathscr{G}^2_{n^{'}_1}$, respectively where $\mathscr{G}^1_{n^{'}_1}$ and $\mathscr{G}^2_{n^{'}_1}$ are the $k \times 1 : n^{'}_1$ part of the generator matrix $\mathscr{G}^1_n$ and $\mathscr{G}^2_n$, respectively. Therefor the remaining $(n_2 - 2n^{'}_1)$ bits are received in phase 2 from both relays and sources. In phase 2 the transmitted codewords by S_1 and S_2 are given by $x_{12'} = u_1\mathscr{G}^{12}_{n^{'}_2}$ and $x_{22'} = u_1\mathscr{G}^{21}_{n^{'}_2}$, respectively where $\mathscr{G}^{12}_{n^{'}_2}$ and $\mathscr{G}^{22}_{n^{'}_2}$ are the $k \times n^{'}_1 + 1 : n^{'}_1 + (n_2 - 2n^{'}_1)/4$ part of the generator matrix $\mathscr{G}^1_n$ and $\mathscr{G}^2_n$, respectively. Therefore, in protocol B, the code word transmitted by S_1 and S_2 are $x^{'}_1 = [x_{11'}x_{12'}] = u_1\mathscr{G}^1_{n'}$ and $x^{'}_2 = [x_{21'}x_{22'}] = u_2\mathscr{G}^2_{n'}$, respectively where $\mathscr{G}^1_{n'}$ and $\mathscr{G}^2_{n'}$ are the $k \times 1 : n^{'}_1 + (n_2 - 2n^{'}_1)/4$ part of the generator matrix $\mathscr{G}^1_n$ and $\mathscr{G}^2_n$, respectively. The received codeword at destination from r_1 and r_2 are $y_1^{'} = \alpha_{11}u_1.\mathscr{G}_{n'}^{3} \oplus \alpha_{12}u_2.\mathscr{G}_{n'}^{4}$ and $y_2^{'} = \alpha_{11}u_1.\mathscr{G}_{n'}^{5} \oplus \alpha_{12}u_2.\mathscr{G}_{n'}^{6}$ respectively where the length of generator matrix $\mathscr{G}_{n'}^{i}, (i = 3, 4)$ is $2k \times (n_2 - 2n_1^{'})/4$ and the length of generator matrix $\mathscr{G}_{n'}^{i}, (i = 5, 6)$ is $2k \times (n_2 - 2n_1^{'})/4$. The code in Equation 3.19 can be viewed as an integrated code with packets $[u_1\ u_2]$ and generator matrix $\mathscr{G}'$ that is given bellow:

$$\mathscr{G}' = \begin{bmatrix} \mathscr{G}_{n'}^{1} & 0 & \alpha_{11}\mathscr{G}_{n'}^{3} & \alpha_{21}\mathscr{G}_{n'}^{5} \\ 0 & \mathscr{G}_{n'}^{2} & \alpha_{12}\mathscr{G}_{n'}^{4} & \alpha_{22}\mathscr{G}_{n'}^{6} \end{bmatrix}. \tag{3.20}$$

Then, decoding of the rateless code is done based on the generator matrix $\mathscr{G}'$. After propagating over a noisy channel, the transmitted code word $x_1^{'}$, $x_2^{'}$, $y_1^{'}$, and $y_2^{'}$ are received at destination as $\hat{x_1^{'}}$,$\hat{x_2^{'}}$, $\hat{y_1^{'}}$ and $\hat{y_2^{'}}$. Let $c = [x_1^{'}x_2^{'}y_1^{'}y_2^{'}] = \{c_1, c_2....\}$ is the transmitted code vector and $c' = [\hat{x_1^{'}}, \hat{x_2^{'}}, \hat{y_1^{'}}, \hat{y_2^{'}}] = \{c_1^{'}, c_2^{'}, ...\}$ is the received code vector at the destination over a noisy channel where $c_o^{'}$ is channel output of transmitted bit c_o .

In noisy channel, the decoding of fountain code is accomplished using the standard BP algorithm on generator matrix $\mathscr{G}'$. At the l^{th} decoding attempt, it performs BP decoding on generator matrix $\mathscr{G}'$ by iteratively passing the LLR (log-likelihood ratio) messages from input bits to output bits, and then from output bits back to input bits. Let $\mu_{c_o,v_i}^{j,l}$ and $\nu_{v_i,c_o}^{j,l}$ denote the message passed from the output bit c_o to the input bit v_i and input bit v_i to the output bit c_o respectively at the j^{th} iteration of l^{th} decoding attempt. In every iteration, the following message update rules are applied in parallel to all input and output nodes in the factor graph

$$tanh\frac{\mu_{c_o,v_i}^{(j,l)}}{2} = tanh\frac{(Z_{c_o})}{2}\prod_{i' \neq i} tanh\frac{\nu_{v_i,c_o}^{j,l}}{2}, \tag{3.21}$$

$$\nu_{v_i,c_o}^{(j+1,l)} = \sum_{o' \neq o} \mu_{c_{o'},v_i}^{(j,l)}, \tag{3.22}$$

where Z_{c_o} is log-likelihood ratios (LLR) of the output bit c_o that is calculated based on the channel observation and knowledge of the CSI at the receiver. We use binary phase shift keying (BPSK) as modulation scheme and assume that the transmitted codeword $c_o \in 0,1$ is equal probability. Therefore, in the Rayleigh fading channel while channel state information is available at the receiver, the log likelihood ratio corresponding on the output node c_o can be expressed as

$$Z_{c_o} = ln\left(\frac{P(c_o = 0|c'_o)}{P(c_o = 1|c'_o)}\right) = \frac{2}{\sigma^2}c'_o.a, \tag{3.23}$$

where a is the normalized Rayleigh fading factor with $E[a^2] = 1$ and density function $f(a) = 2a\ \exp(-a^2)$. In the end of l^{th} decoding attempt, if the destination is confident that the transmitted packets u_1 and u_2 are decoded successfully, it then sends an ACK through a noiseless feedback channel to the sources and relays to terminate the transmission of the current code words. Otherwise it collects more output symbols from sources and relays and initiates next decoding attempt to decode again.

3.6 Simulation Results

In this section, we conduct simulations to investigate the performance of JNFC in Rayleigh fading channel. Two sources S_1 and S_2 generate original packets u_1 and u_2 where each of length $k = 500$ bits. Using LT codes then the original packets u_1 and u_2 are encoded into x_1 and x_2 respectively and transmitted to the destination. After correctly decoding the source packets, relay nodes re-encode the packets using a suitable degree distribution and perform network coding on them. We assume that both sources have the same distance to the destination and the relay nodes are more closer to the destination than the sources. We also assume that both relays have the same destination to the destination and the SNR between relays and destination is higher than the SNR between sources and destination. The SNR of relay-destination link is given by $SNR_{rd} = SNR_{sd} + 10dB$ where SNR_{sd} is the SNR of source-destination link. In this work, we propose two JNFC schemes depending on the coding techniques at relay nodes such as:

- JNFC-RLNC: In this case, relay nodes re-encode source information using LT codes. Then the coded symbols of two packets are randomly chosen to generate network coded symbols. However, in this way the resulting network coded symbols do not follow Robust Soliton Distribution in degree.
- JNFC-MLT: In this scheme, both sources and relays transmit code symbols that follow RSD as in degree distribution. After receiving source packets, relay nodes use DSD to encode each source information. The coded symbols of two sources are then selectively combined in such a way that the result in code symbols follow the degree distribution of LT codes.

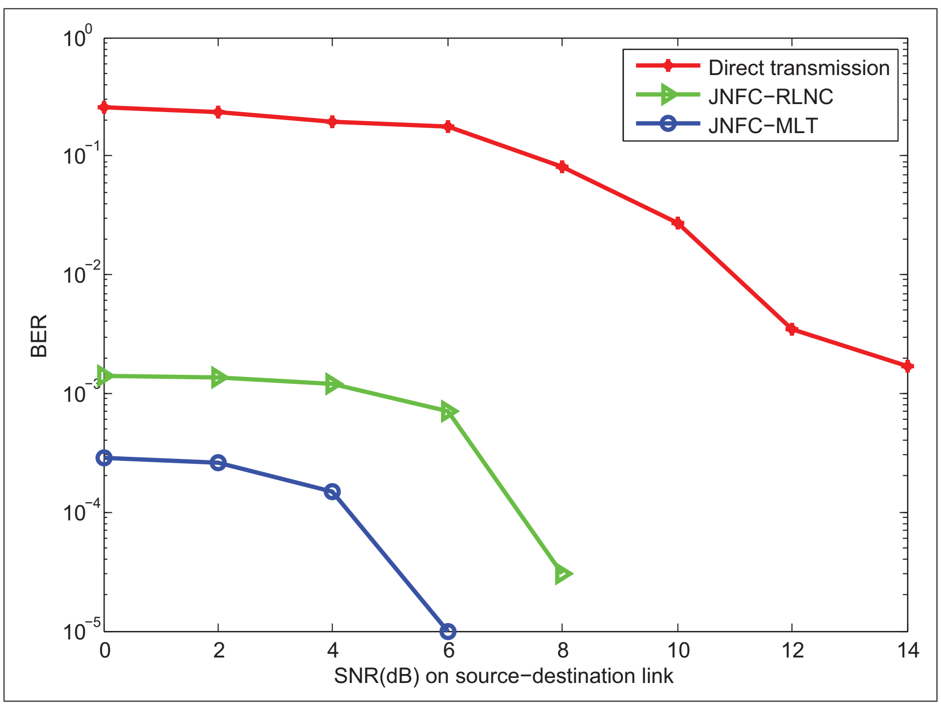


Figure 3.8 Bit error rate Vs SNR (dB)

Figure 3.8 presents the bit error rate (BER) of various schemes over the signal to noise ratio (SNR) on the source-destination link. The decoding failure probability at the decoder, δ is considered as .5. and LT design parameter λ is considered .1. The number of received symbols from each source and relay is considered as $k.\beta$. It has been observed that JNFC schemes significantly improve BER performance. This is due to the redundancy and diversity provided by the use of relay nodes and the use of network coding. Moreover it is shown that JNFC-MLT outperforms JNFC-RLNC . The performance of JNFC-MLT is about more than 3 dB compare to JNFC-RLNC for a BER of 10^{-4}.

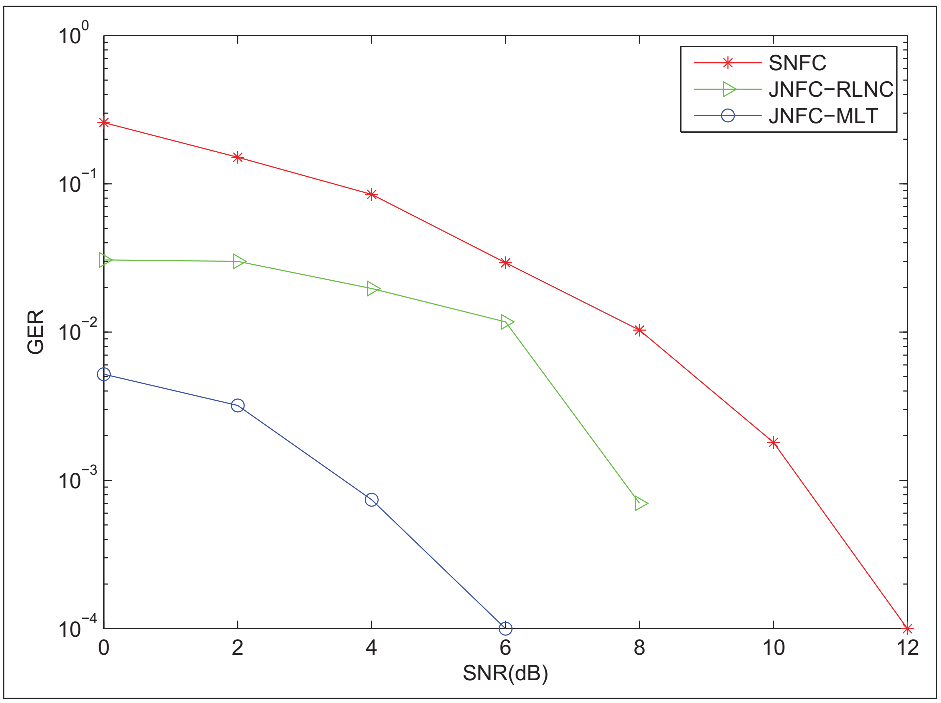


Figure 3.9 Generation error rate Vs SNR (dB)

Figure 3.9 compares the different JNFC schemes and separated network and fountain coding (SNFC) in terms of generation error rate (GER). Two packets u_1 and u_2 are generated at each generation. The generation error occurs when at least one of the two packets can not be recovered at the destination correctly. It is shown that SNFC has a performance loss of around 3 dB and 6dB compared to the JNFC-RLC and JNFC-MLT , respectively for a GER of 10^{-3} . This is because in JNFC the redundancy both in channel coding and network coding are efficiently exploited. However, in SNFC the packets that fail channel decoding can not exploit the redundancy packet transmitted by the relay nodes.

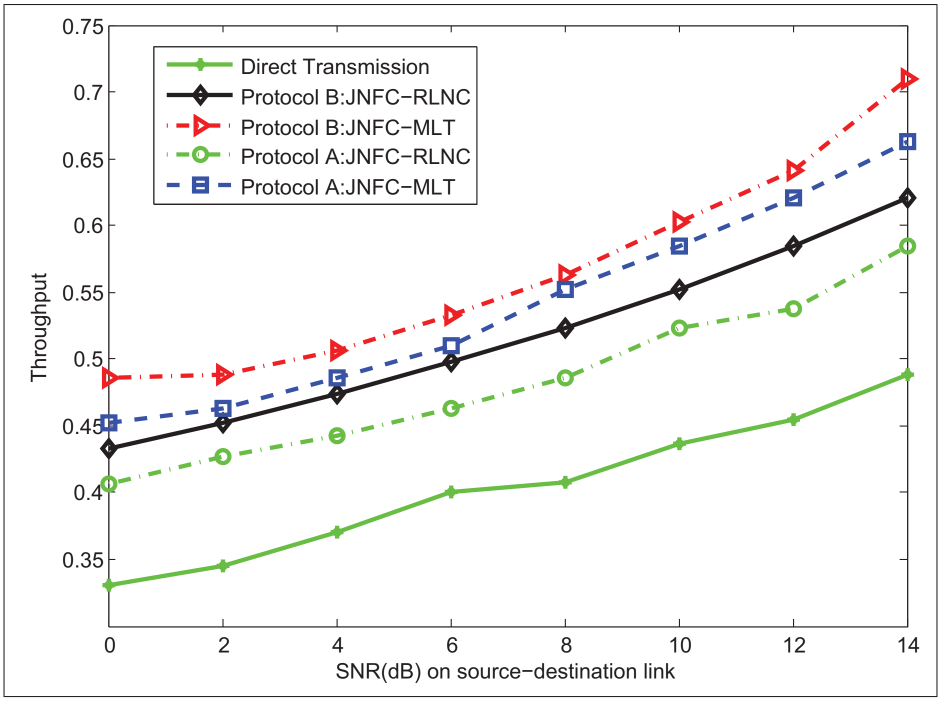


Figure 3.10 Throughput Vs SNR (dB)

Figure 3.10 compares the performance of proposed protocols using different JNFC schemes. In protocol A, source nodes cease its transmission when both relay nodes successfully decode their information. After that relay nodes continue their transmission until they receive an ACK from destination that the decoding is successful. On the other hand in protocol B, source nodes and relay nodes continue their transmission until they receive an ACK from destination that the decoding is successful. Throughput is calculated by the following equation:

$$\eta = \frac{Number\ of\ received\ symbols\ for\ successful\ decoding}{Number\ of\ original\ symbols}. \tag{3.24}$$

We compare the performance of JNFC with the direct transmission where source nodes transmit to destination directly without any help of relay and continue their transmission until they receive ACK from destination. It has been shown that JNFC schemes significantly improve the throughput performance than the direct transmission. Moreover, it has been noticed that JNFC-MLT outperforms JNFC-RLNC regardless of the employed protocol. Moreover, the performance gap between this two JNFC schemes more apparent in Scenarios A. This is because in JNFC-MLT, the transmitted symbols by relay nodes are actually like LT codes both in performance and degree distribution.

3.7 Conclusion

In this chapter we developed several joint network and fountain coding schemes (JNFC) for reliable communication in wireless networks and evaluate the performance of the proposed schemes under different transmission protocols. JNFC seamlessly combines fountain and network coding techniques and thus efficiently use the redundancy. Simulation results demonstrate that JNFC can greatly improve the performance of the system when compared to separated network and fountain coding scheme and direct transmission. In addition, simulation results demonstrate that the encoding schemes at relay nodes play a significant role to improve the performance of the system. Regardless of the protocols JNFC-MLT outperforms JNFC-RLNC in throughput performance.

CHAPTER 4

JOINT NETWORK CHANNEL FOUNTAIN SCHEMES FOR RELIABLE UPLINK TRANSMISSION OVER LTE-A CONNECTED M2M NETWORK

4.1 Introduction

Machine-type communication (MTC) enables a wide variety of applications from mission-critical services to massive deployment of autonomous devices. This massive devices will be planned for diverse applications with wide range of requirements and characteristics. To spread these applications widely, cellular networks are considered as a potential candidate to provide connectivity for MTC devices. However, cellular network has been generally optimized for human-to-human (H2H) communications such as voice or web browsing and bandwidth demanding services where the amount of uplink traffic is normally lower than the downlink traffic. The traffic characteristics and requirements for M2M applications are very much different from human-to-human (H2H) communications. In major M2M applications, small and infrequent data is generated from a mass number of devices. High reliability and end-to-end latency of a few ms are the requirements of many future M2M applications such as wireless automation and control. Current state-of-the art technology, such as LTE-A, is clearly far away from meeting these requirements. Therefore, these requirements must be effectively addressed for the future broadband wireless communications toward 5G systems in order to fully support MTC. In this vein, LTE has already launched numerous activities to support MTC for future releases of LTE network. Though various attempts have been made by LTE to support M2M communications the LTE architecture and protocols can still be improved for reliable massive M2M communications.

One of the proven ways of improving reliability of data transmission over cellular wireless networks is hybrid automatic repeat request (HARQ) mechanism. In particular, LTE employs a rate 1/3 Turbo code with a contention free quadratic permutation polynomial (QPP) as forward error correction (FEC) coding at the physical (PHY) layer, and the automatic repeat request (ARQ) at the medium access control (MAC) layer to improve the reliability of data transmission. The ARQ part consists of a stop and wait (SAW) process, where the transmitter stops and waits after each transmission to receive an acknowledgment (ACK) or negative ACK (NACK) from the receiver. In either case (ACK or NACK), the transmitter is required to schedule and process the next transmission within a specific time period. If the received data has an error then the receiver buffers the data and requests a re-transmission from the source. When the receiver receives the re-transmitted data, it then combines it with buffered data prior to channel decoding and error detection. This helps the performance of the re-transmissions. However the above mentioned technique may increase the delay to unacceptable levels when more than one retransmission are require to correctly decode the source information. In order to optimize throughput and achieve high peak data rate, LTE takes advantage of adaptive coding and modulation (AMC) in addition to HARQ process. When AMC is implementing with HARQ, the modulation and coding scheme (MCS) is adapted between each retransmission that significantly enhance the performance (Villa *et al.*, 2012). However, it requires sophisticated channel estimation schemes in order to obtain channel state information (CSI) and to provide channel quality indicator (CQI) feedback for the adaptation mechanisms. In the presence of large amount of devices in the network, the transport of CQI feedback may result in a significant amount of control signaling overhead. Moreover, CQI could be unavailable or outdated at the eNB due to the sleep mode mode (e.g, DRX and eDRX mechanism for power saving of the M2M devices) and moderate to high mobility of the M2M devices.

In contrast to typical fixed rate code like Turbo code where the block length is set before the transmission, fountain code (Luby, 2002; Shokrollahi, 2006; Byers *et al.*, 1998/10/) is a rateless code. In fountain coding, the source unconscious of channel state information (CSI) can generate as many encoded symbols as necessary on the fly by simply performing module 2 operations on the source symbols and continues its transmission until it receives an ACK from the receiver. In multiple access transmissions, fountain codes effectively use the collided packets in an effective way to increase the overall system throughput. As a promising feature, they do not require any synchronization mechanism as virtual MIMO and precise CSI/CQI feedback at the transmitter as Turbo codes. This features are particularly advantageous for

M2M communications. Since a large number of devices are installed in a cell so it would be impractical for the eNB to obtain CSI/CQI to every MTC device (Louie *et al.*, 2012). Also the transport of CQI feedback will result control overhead. Moreover, for various reason the CQI could also be unavailable or outdated. As for example to reduce power consumption, devices that send sporadic traffic are encouraged to go to sleep mode in DRX and eDRX mechanism. Hence CQI information may be outdated since the device is unable to promptly report it during sleep mode. In such cases fountain code could be a good solution for uplink transmission between massive MTC devices and LTE eNB. In (Soljanin *et al.*, 2006), the author showed that the punctured incremental redundancy (IR-HARQ) scheme based on LDPC codes performs better on high SNR region while rateless IR-HARQ scheme based on Raptor codes perform better in low SNR region. Recent study(Chen *et al.*, 2015) showed that Raptor code based HARQ scheme achieves a similar performance to the LTE Turbo-coded HARQ scheme or even outperforms the LTE Turbo-coded HARQ in block fading channel.

Motivated by the performance of the fountain code in previous works (Soljanin *et al.*, 2006; Chen *et al.*, 2015), in this chapter we developed a set of joint network and fountain coding schemes for reliable uplink transmission between MTC devices (MTCD) and LTE-A base station(ie, eNB) (Nessa & Kadoch, 2016). In a large M2M network, when a massive number of nodes will attempt to connect to an LTE-A base station directly, it is very likely that many devices will try to access the random access channel (RACH) at the same time. This would cause many random access opportunity (RAO) collisions, leading to a waste of resources and device energy. One possible solution for this problem is the use of clustering, where the devices are grouped into number of clusters and more powerful nodes have been chosen as cluster heads (CHs). The machines belonging to a cluster communicate to CH, which then aggregates the traffic and forwards to eNB directly or with the help of the LTE-relay. We investigate M2M communications over type I and type II relay enhanced LTE-A networks. In type I relay enhanced networks, CHs sends their data to the relays using LT codes. After successfully decoding the information of CHs, relays generate new symbols from received symbols using network and fountain coding. Relays then synchronize with each other and transmit the encoded symbols using the Alamouti space-time codes (Alamouti, 1998) to the eNB. In type II relay enhanced LTE-A networks, CHs encode their information using LT codes and then broadcast their information in the system until relays successfully decode their messages. After correctly decoding of the source messages each relay generates new message from received messages using network and fountain coding and then forwards to the eNB. Both relays continue their transmission until eNB successfully decodes both messages. We evaluate the performance of the proposed schemes in terms of bit error ret (BER) and transmission efficiency. However the performance comparison of our proposed JNFC schemes with HARQ and LTE-tubo coded HARQ is left for our future work.

This remaining of this chapter is organized as follows. We first present a brief discussion about LTE Turbo encoding procedure, LTE uplink data transmission scheme and LTE uplink HARQ process in Section 4.2. We review the sate of art solutions that have been proposed to increase reliability in M2M communications in Section 4.3. We then present the proposed JNFC scheme using a two-source two-relay network and corresponding encoding, decoding, and calculation of transmission efficiency in Sections 4.4and 4.5, respectively. Finally, Section 4.6 demonstrates the simulation results, followed by Section 4.7 to conclude this chapter.

4.2 Background

4.2.1 LTE Turbo Code and Circular Buffer

LTE-A uses 1/3 rate Turbo code as a channel coding scheme at the physical layer. For an input block size of K bits, the output of a Turbo encoder consists of three length-K streams, corresponding to the systematic bit and two parity bit streams referred to as “Parity 1”, and “Parity 2” streams as well as 12 tail bits due to trellis termination. Therfore, the actual mother code rate is slightly lower than 1/3. In LTE, the tail bits are multiplexed to the end of the three streams, whose lengths are hence increased to $(K+4)$ bits each. Each of the three output streams of the Turbo coder is rearranged with its own sub-block interleaver as shown in Figure 4.1. Then, a single output buffer is formed by placing the rearranged systematic bits in the

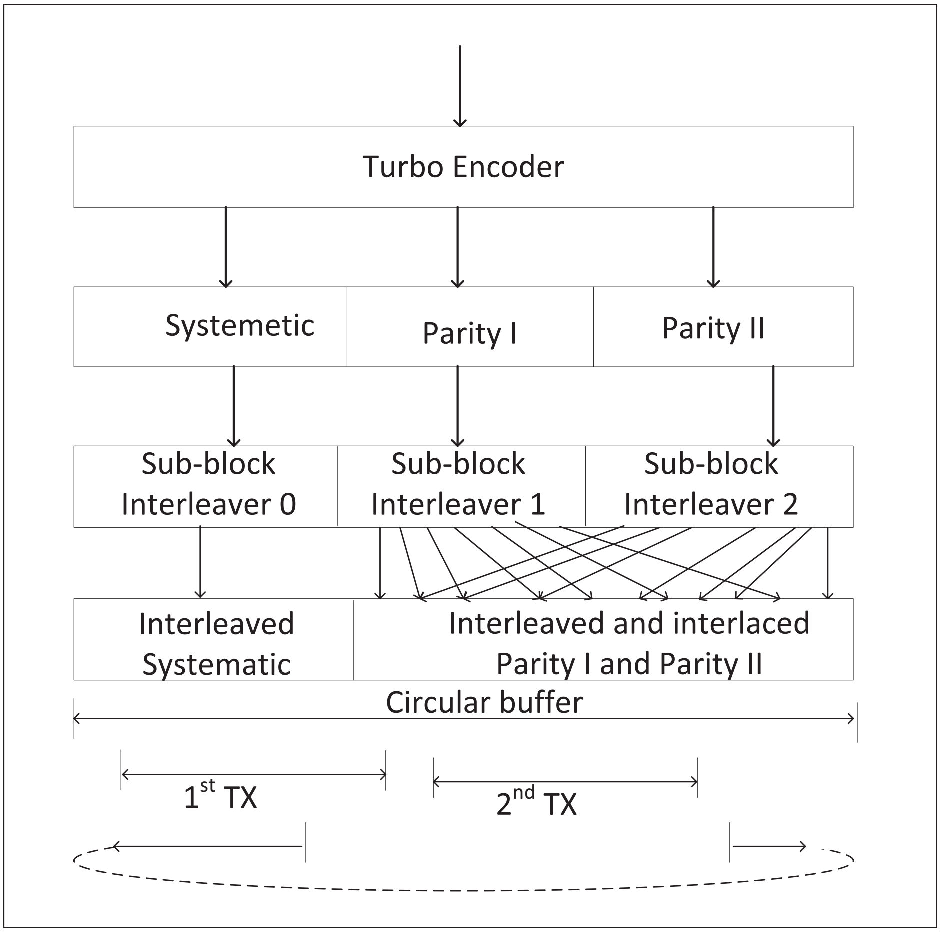


Figure 4.1 Operations of circular buffer rate matching for turbo code

beginning followed by bit-by bit interlacing of the two rearranged parity streams. At the first transmission, the systematic part of the output buffer and a selected number of parity symbols are sent. If the decoding fails then the receiver buffers the data and requests a re-transmission from the source. The transmitter sends set of additional parity symbols from the codeword of the mother code. An rate matching (RM) algorithm is used to generate different redundancy version(RV) of the codeword for different transmission. For example for high code rates only some portion of the circular buffer is transmitted whereas for the very low code rate circular buffer can be transmitted several times. So, in general for any required transmission code rate the circular buffer can be read out from an arbitrary starting point. If the end of the buffer is reached but the receiver does not decode successfully then it overlaps from the beginning of the buffer, hence named as circular buffer. When the receiver receives the re-transmitted data, it then combines it with buffered data prior to channel decoding and error detection.

4.2.2 LTE Uplink Transmission Scheme

LTE uses asymmetric multiple access schemes in the downlink and uplink. The multiple access scheme in the downlink is based on orthogonal frequency division multiple access (OFDMA). Due to its multi-carrier nature, OFDMA is compatible for providing high peak data rates in high spectrum bandwidth. However, OFDMA technique results in high peak-to-average power ratio (PAPR) of the signal, which leads to low power efficiency and makes it less favorable for the uplink transmission. Hence, LTE uses single-carrier FDMA technique (SC-FDMA) as the multiple access scheme for uplink, which is much more power efficient than OFDMA. This technique is also known as discrete fourier transform (DFT)-spread OFDM (DFTS-OFDM). The SC-FDM modulation is quite similar to the OFDM except that before the Inverse DFT (IDFT) in transmission side of OFDM, an extra DFT processing is added in the DFTS-OFDM and vice versa after the DFT in OFDM receiver side.

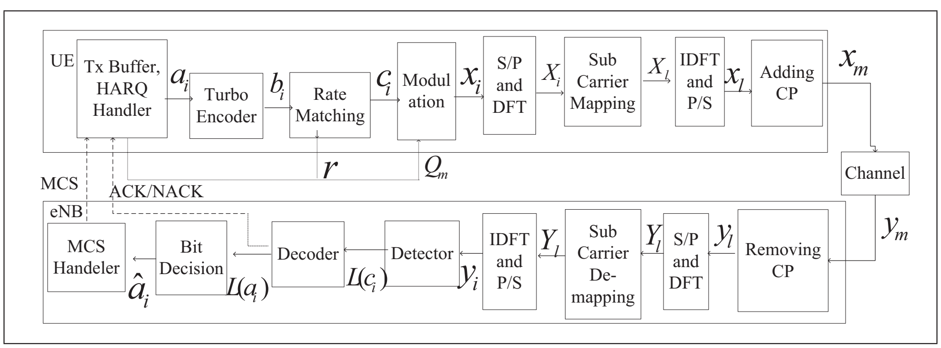


Figure 4.2 LTE uplink transmission and reception with HARQ transmission principle

Figure 4.2, shows the block diagram of LTE uplink transmission and reception. First the data sequence a_i of length N_a is encoded by Turbo encoder. The output of the Turbo encoder is a codeword sequence b_i of length N_b. This overall codeword has the mother code rate $Na/Nb = 1/3$. The encoded codeword, is then passed to a rate matcher, which selects a number of bits depending on the redundancy version (RV) r and the number of available physical resource block n_{PRB} to construct the codeword c_i of length N_c. The rate matched codeword is then passed to baseband modulator to form a vector x_i of modulated complex symbols. LTE uses adaptive base band modulation scheme, thus depending upon the channel it adopts the modulation formats. The common modulation being used in LTE are quadrature p shift keying (QPSK), binary phase shift keying (BPSK), 16 level quadrature amplitude modulation (16-QAM) and 64-QAM. The code bits $c_{i,l}$ of the codeword c_i with $l = 1,...,N_c$ are mapped to symbols $x_{i,k}$ of x_i where $k = 1,2,...N_m$. The next step is to convert the serial modulated symbols, x_i of length N_m into N_m parallel data streams. Then it performs an N_m point DFT, this step will transform time domain modulated symbols x_i to frequency domain symbols X_i. Next, the transmitter maps the outputs of the DFT block to N_{total} orthogonal subcarriers. The result of the subcarrier mapping is the set X_l complex subcarrier amplitudes of length N_{total}. After subcarrier mapping, an N_{total} point Inverse DFT (IDFT) operation is performed to generate a time domain signal x_l. Finally, after adding cyclic prefix (CP) and passing through the transmission filter for pulse shaping, the signal x_m is transmitted through the channel.

At the receiver, an SC-FDMA receiver does exactly the inverse of what have been done at transmitter. First receiver remove the CP. After removing the CP the receiver transform the received signal into frequency domain with the help of DFT. Afterwards, subcarrier de-mapping is done following the same method in which subcarrier mapping was done in the transmitter. Next, an equalizer compensates for the distortion caused by the multipath propagation channel. After the equalization process, the IDFT block transforms the signal into the time domain and detection is done in time domain . Let the input of the detector is the symbol vector y_i. After receiving the symbol $y_{i,k} \in y_i$ a, the receiver calculates the log-likelihood ratio $L(c_i,l)$ of the participating coded bit c_i,l. These LLRs are then passed to the Turbo decoder, which delivers decoded LLR values for the information bits $L(a_i,k)$. After a hard decision based on these LLRs a cyclic redundancy check (CRC) is done. If the CRC fails a NAK will be transmitted to UE to denote a decoding failure, otherwise an ACK is transmitted over the feedback link.

4.2.3 LTE Hybrid Automatic Repeat Request (HARQ) Scheme

In LTE/LTE-A, HARQ is utilized as a stop and wiat (SAW) process where it has to wait for feedback before the next transmission or retransmission. In LTE FDD there are 8 SAW process. When a transmission has been made from a particular process, it waits for an ACK/NACK. Till it receives ACK/NACK, the process will be in-active state and will not process another transmission. Each transmission in LTE corresponds to 1 transmission time interval (TTI) and each SAW process should process the data within 1 ms or 1 sub frame. Since there are 8 SAW process hence each process will have to wait for 8 ms before re-transmission over the air interface. Therefore, more than one retransmission may increase the delay to unacceptable levels. In order to optimize throughput and achieve high peak data rate, LTE uplink use adaption techniques to adapt the actual transmission parameters to the link quality, which can be separated into an outer and an inner link adaptation (LA), as shown in Figure 4.2. The inner loop link adaptation (ILLA) takes the first responsibility to select the suitable MCS for the UE. To determine the MCS, the eNB measures UL SNR of the reference symbols. If the decoding fails in one HARQ transmission the inner LA is used, which is based on a simple ACK/NAK feedback. If a NAK is received at the UE, the redundancy version r is changed and another codeword $c_(i+1)$ is selected based on r. Channel Quality Indicator (CQI) is an important parameter to set the value of r. The CQI measurement interval, measurement resolution in frequency domain, reporting mechanisms are all configurable parameters and have a tremendous impact on the system performance. For various reasons the ILLA can not always provide the perfect adaptation and thus outer loop link adaptation (OLLA) function is needed. The function of the OLLA is to adapt the MCS selection to provide certain block error rate (BLER). In the outer loop, eNB informs the user equipment (UE) about the modulation coding scheme (MCS)that will satisfy the maximum block error rate requirement. In order to perform the link adaptation, the transmitter requires to obtain channel side information (CSI) to adapt its transmissions. Hence, a feedback channel is utilized to carry CSI to the transmitter. This channel can also carry feedback error control information in case of using HARQ scheme.

4.3 Related Work

The use case of M2M communications in the next generation mobile network consortium and 3GPP is divided into two major groups such as massive MTC and mission critical MTC. Massive M2M communications refers to the scenario where a large number of low cost machines/devices (e.g., 10000) are interconnected within a given area and exchange messages with each other. This is relevant for large-scale distributed cyber-physical systems (e.g., smart grid) or industrial control. In this case, the data packets are short and reliability must be high to cope with critical events. Mission critical are a subset of applications within the massive M2M communications category, which requires high reliability and very low latency. In the following we will cover some of the proposed solutions in the existing literature on improving reliability of massive M2M communication over LTE-A networks.

The first step to initiate a data transfer using an LTE/LTE-A network is to perform random access (RA) in a contention manner. Therefore, one problem that could be encountered in the future for massive M2M communications is the overload of the LTE RA procedure under a massive number of access requests. This would cause frequent transmission collisions which leads to network congestion, unexpected delays, packet loss, high energy consumption, and radio resource wastage. Efficient random access is one important aspect to consider for reliability in massive M2M communications. In this vain 3GPP, different academia, industry, and government bodies have proposed several solutions for RA congestion in order to support massive MTC over LTE networks. In (3GPP.37.868, 2013), 3GPP specified six distinct solutions of LTE RA congestion due to massive access request from massive MTCDs. These solutions are access class bearing, MTC-specific back-off, dynamic resource allocation, slotted random access, separate random access resources, pull-based random access. In (Madueño *et al.*, 2014a), the authors proposed a tree splitting RA model to resolve random access collisions rather than just trying to avoid these occurrences. This algorithm works on top of the RACH procedure and is only implemented when a RACH overload occurs. In (Hyder *et al.*, 2014) the author proposed a self-optimization overload control (SOOC) approach that combines RA resource separation, dynamic RA resource allocation, and dynamic access barring scheme. In their work the

MTCDs are subgrouped into two units: high-priority MTCDs and low-priority MTCDs. Under this model, MTCDs send RRC requests along with a counter value which indicates the number of RA attempts they had done before receiving the successful RAR message. By observing the counter value, the eNB estimates the RA congestion level. Depending on the congestion level and available uplink radio resources, the eNB either increases the RA resources, or decreases the access probability of low priority MTCDs, or takes both actions together. Finally, the new parameters of RA resources and access probability are broadcasted to the MTCD. To increase the reliability of M2M communications, some research works have proposed pre-reservation of some resources from the overall bandwidth for M2M communications only. As for example in (Madueño *et al.*, 2014b), the authors proposed resource reservation for M2M application. Then they split theses resources into preallocated and common resources where the preallocated resources are used for sending intended short messages, while the common resources are used for excess messages and retransmission due to transmission errors. In their work all M2M communications are treated equally and are assumed to require resources periodically that does not address the real scenario. As for example, mission critical industrial control applications, communication may not report periodically. Therefore the equal treatment of the M2M devices may be detrimental. In (Abdalla & Venkatesan, 2013) suggested a different method to meet the requirements of different applications of M2M communications. They suggested an M2M-Aware schedular that dynamically allocates a fraction of the available resources for M2M devices according to the requirements of the applications so that the QoE of human users is not degraded. In their work, they proposed a new QoS class identifers (QCIs) to support end to end QoS for M2M communications. The allocated resource of M2M communication are then distributed among the M2M devices based on the M2M specific QCI.

However, the allocation of the limited number of resources orthogonally to devices may be the main bottleneck of slotted-Aloha based random access (RA) for massive MTC in LTE networks. To overcome the problem of HARQ schemes and accommodate a large number of M2M devices some research works have proposed non aloha based RA . In (Wang *et al.*, 2015) the authors proposed a MC-RMA protocol where the the data symbols of each user are spread over a block of REs and each RE can be shared by all the users. Like the the degree distribution profiles of rateless codes, the BS assign some access polynomials to the users. In this work, each user encodes its information bits with an LDPC encoder and then maps the resultant coded bits to a vector of symbols based on BPSK modulation. Then after receiving the access polynomials and the beacon from the BS, each user randomly chooses a certain number of coded symbols, linearly combines them, and then transmits the resultant signal to BS until receiving an ACK from BS. In this way their proposed MC-RMA protocol can adapt the code rate according to channel condition and support massive access. In (Shirvanimoghaddam *et al.*, 2015) the author proposed a RA model based on analog fountain code (AFC) for MTC devices with irregular traffic patterns. AFC-based RA combines multiple access with resource allocation. In this model, the available contention-based RA preambles are sub-grouped based on the delay constraints of MTCDs and each group of RA preambles are assigned to one delay group. The RA process has two phases, i.e. contention phase and data transmission phase.In the contention phase all the MTCDs with the same delay constraints send RA request by using predefined preamble(s)to establish a connection between devices and BS. The BS then estimates the number of contended MTCDs per preamble, and broadcasts this information to all contending MTCDs for each of the detected preambles and transmits a random access response (RAR) message including the number of devices which have selected each RA preamble, the allocated RB for each detected RA preamble, and timing information and optimized access probabilities. Based on the received information from BS each MTCD generates an orthogonal random seed and shares it with the eNB. Therefore, both the eNB and MTCD can construct the same bipartite graph to perform AFC encoding and decoding for subsequent communications . In the data transmission phase, the actual payload data along with the device identity (ID) is transmitted by the devices to the BS unless it receives an ACK from BS.

In recent years improving the current LTE HARQ scheme in order to provide reliable transmission and efficient spectral employment for M2M communications has gain much attention. In the legecy LTE-HARQ scheme if one or more retransmission are required to ensure a successful decoding, then 8 ms of delay is incurred for every RTX. In poor channel condition, this makes the current LTE HARQ process unsuitable for delay constraints MTC applications due to the high probability of number of retransmission. For this reason, in (Osseiran *et al.*, 2014) the authors recommended to support only one retransmission to implement HARQ scheme for ultra reliable communications with low-latency. In (Qian *et al.*, 2013), Qian et al proposed a adaptive HARQ protocol for MTC known as LTE-M adaptive HARQ where the coding rate of initial frame transmission is determined by the channel quality information (CQI). The LTE-M adaptive HARQ scheme works on the basis of the original Turbo channel code in the LTE, which has an initial coding rate of 1/3. With the aid of puncturing operations, this 1/3-rate Turbo encoded rate may match any coding rate higher than this. During the first transmission, the coding rate is adapted based on the CQI. At the first decoding attempt, if the Turbo decoding succeeds, an ACK message is sent back to the transmitter. If the transmitter receives an NACK it then sends the whole 1/3-turbo encoded packet for the second transmission. At the receiver, chase combining will be applied to those repeated bits during both transmissions. Then, the Turbo decoding is activated again. If the CRC is satisfied after the decoding, an ACK message is fed back to stop the transmission of this packet. If it fails again, the packet will be discarded. Only two transmissions are allowed in the LTE-M HARQ. In legacy LTE HARQ scheme incremental redundancy (IR) used to determine the size of the retransmission. In (Woltering *et al.*, 2014) the author focuses on an adaptive retransmission scheme which configures the retransmission size by using a reliability based approach instead of using fixed IR scheme. Based on channel SNR information and bit error probability information from the decoder output, they adapt the retransmission size in terms of physical resource blocks. In (Cabrera, 2015) the author proposed an Adaptive HARQ (A-HARQ) for mission critical industrial control and medical applications. For a quick retransmission in this work, the sender pre-constructs and pre-encodes different redundancy version (RV) with different combinations of systematic bits and parity bits, and storing them in the buffer. They used channel quality indicator (CQI) reports to select the best sub-bands to make the retransmission successful. Moreover they utilised TTI bundling to increase the number of retransmission within a 4 ms time period. It is shown that A-HARQ incurs 35% less delay than the legacy HARQ, with a slight decrease in throughput. However the above mentioned adaptive HARQ schemes can boost the reliability for ultra reliable communications (URC) if the feedback channel provides accurate CSI and delivers ACK/NACK messages without error. The unsuccessful decoding of ACK messages affects the reliability as well as results in unnecessary data transmissions that consumes radio resources. On the other hand, the unsuccessful decoding of NACK messages reduces the reliability.

4.4 System Model

To accommodate large M2M traffics and reduce the random access delay under limited random access resources, we consider clustered heterogeneous M2M network structure. We assume the network is partitioned into number of clusters and more powerful nodes have been chosen as CHs. Within clusters MTC devices communicate with each other by using short range communication technology. We also assume that CHs are more powerful nodes with long range communication technologies that collect data from their cluster members, carry out the necessary translation (or encapsulation) between short range communication technology and LTE and, finally, forward the aggregated data to eNB directly or with the help of relay nodes. Obviously the power consumption at CH is more higher than the ordinary cluster members. Therefore, different energy transmission techniques (Kurs *et al.*, 2007; Karalis *et al.*, 2008) may apply to prolong the lifetime of the network.

4.4.1 Network Scenario:Type I Relay Enhanced LTE-A Network

In this scenario MTC devices are located at outside of the coverage of eNB and type I relays are deployed to extend the coverage of macrocell as shown in Figure 4.3. The information block of each source is denoted as m_k with a length of M_k, where $k = 1,2$. Each source S_k generates a large number of code stream using LT codes. The code stream is then modulated and sequentially transmitted to relays in phase I. Source CHs continue their transmission until relays successfully decode their messages. After successfully decoding the source messages each R_i concatenates the message m_1 and m_2 and perform LT coding onto the unified message. In phase II, the output symbols of R_1 and R_2 are synchronized over time to form consecutive

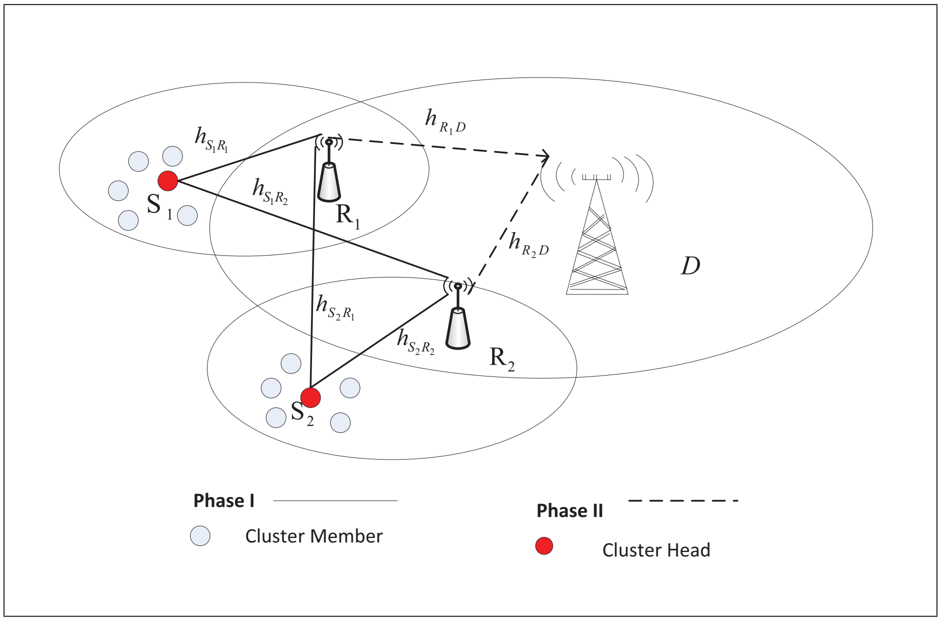


Figure 4.3 M2M communications over type I relay enhanced LTE-A network

pairs of symbols and transmit sequentially according to the Alamouti space-time code to the eNB. The eNB receives the space-time code (STBC) and performs standard Alamouti decoding. It is assumed that relays either use an identical random-number generator in LT coding or share the degree distribution of the LT code among themselves.

4.4.2 Network Scenario: Type II Relay Enhanced LTE-A Network

The UL cooperative communications framework in a type II enhanced LTE-A network is shown in Figure 4.4 where an eNB D is located in the centre of the cell and CHs S_1, and S_2, intend to connect to the eNB simultaneously for UL transmission. Near the cell edge two type II relays (RS) , R_1 and R_2 are installed to assist the uplink transmission and improve the transmission rate of the users in the cell. In the phase I, each source S_k generates a large number of code stream using LT codes. The code stream is then modulated and sequentially transmitted to the eNB and relays. Both eNB and relays make decoding attempts to recover source messages. Since the source-relay link is better than that of the source-eNB link, the relays can almost always successfully decode before the destination does. As soon as relays

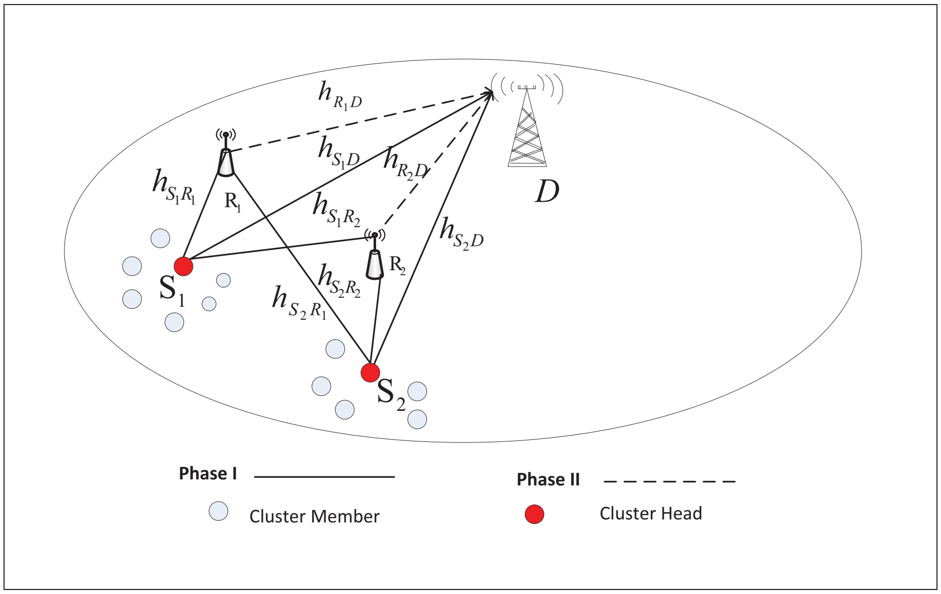


Figure 4.4 M2M communications over type II relay enhanced LTE-A network

have acquired sufficient information to decode sources data, they transmit an acknowledgment to the sources that their reception was successful. In phase II, they generate new codestreams from the sources data using network and fountain coding and transmit to the destination to provide additional error protection. Both relays continue their transmission until D successfully decodes both messages.

4.4.3 Channel Model

For simplicity, we assume that all nodes are operated in half-duplex mode, i.e., they can either transmit and receive at a time. The propagation channels between different nodes are independent and identically distributed (i.i.d.) Rayleigh fading with additive white Gaussian noise (AWGN). We adopt binary phase shift keying (BPSK) modulation thus the channel model between any transmitter i and receiver j can be written as:

$$y_j[n] = H_{i,j}x_i[n] + n_j[n] \tag{4.1}$$

where $x_i[n]$ is the BPSK modulated transmitted symbol, $y_j[n]$ is the received symbol, $n_i[n]$ is zero mean and additive white Gaussian noise with variance σ_i^2 at time instant $[n]$, $H_{i,j}$ is the channel gain that is modeled as $\sqrt{G_{i,j}}h_{i,j}$ where $h_{i,j} \sim \mathcal{N}(0,1)$, a circularly symmetric complex Gaussian random variable with zero mean and unit variance, and $G_{i,j}$ denotes the path loss between node i,and node j. Without loss of generality, we assume a unit path loss between each source and destination , i.e., $G_{S_k,D} = 1, k = 1,2$. We then have $G_{S_k,R_i} = \left(\frac{d_{S_k,D}}{d_{S_k,R_i}}\right)^{\alpha}$ and $G_{R_i,D} = \left(\frac{d_{S_k,D}}{d_{R_i,D}}\right)^{\alpha}$ where d_{S_k,R_i} and $d_{R_i,D}$ are the distance of S_k and D from R_i , respectively and $\alpha = 2$. Therefore, the signal-to-noise ratio (SNR) between node i and j is given by

$$\gamma_{i,j} = \frac{P_i|H_{i,j}|^2}{\sigma_i^2} \tag{4.2}$$

where P_i denotes the transmitted power of node i. We assume orthogonal channels (Zhu & Wang, 2009, 2012) are assigned to different terminals. The sources transmit to the destination and relays over different orthogonal channels and the relays transmit to the destination over other different orthogonal channels. Moreover, we assume that all the nodes use their own fountain coding routine to encode the data and the encoding degree distribution of the transmitted packets are sent to the destination eNB by an extra robust CDMA channel with long sequence length instead of using identical random-number generator both in encoder and decoder as used in (Liu & Lim, 2009), that needs strict synchronization and increases the complexity in multiple relay network.

4.5 JNFC Schemes for a Two Source Two Relay Network

4.5.1 Encoding Scheme at Sources

We assume each source wants to transmit k bit of information to destination D. Source S_k encodes its information block m_k into codeword $\{x_k = x_k(1), x_k(2), \ldots . x_k(n) \ldots\}$ using LT codes. The set of encoded symbols generated by S_1 is given by

$$x_1 = m_1 \mathscr{G}_n^1, \tag{4.3}$$

where, $\mathscr{G}_n^1$ is the generator matrix of the code symbols that are truncated to length n. Similarly, the set of encoded symbols generated by source S_2 is given by

$$x_2 = m_2 \mathscr{G}_n^2, \tag{4.4}$$

where, $\mathscr{G}_n^2$ is the generator matrix of the code symbols that are truncated to length n. Then x_1 and x_2 are BPSK-modulated and broadcasted into the system using different orthogonal channels. Source S_1 and S_2 continues their transmissions until their information block m_1 and m_2 are correctly recovered at R_1 and R_2.

4.5.2 JNFC Scheme at Relays

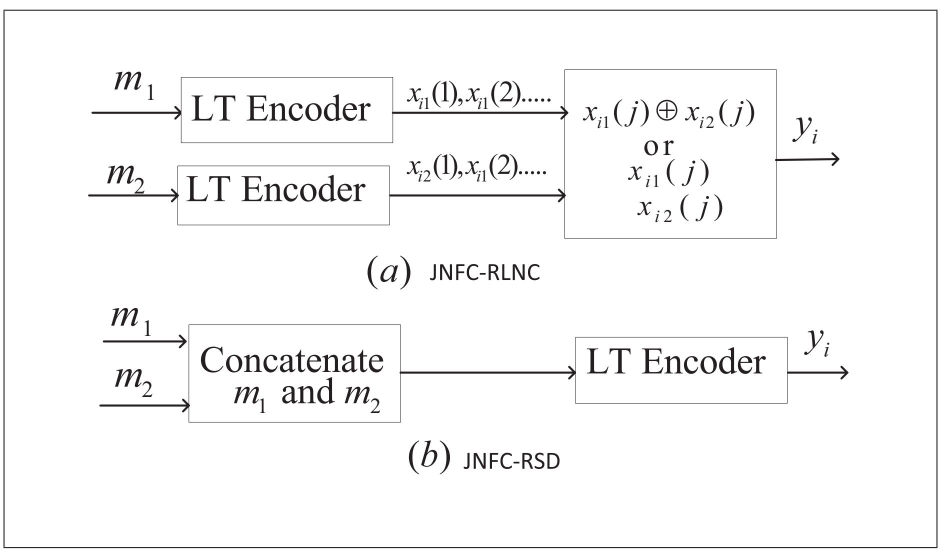


Figure 4.5 Encoding schemes at relay nodes

After correctly decoding information blocks of both sources, R_i generates new code y_i using network and fountain coding from the received information blocks m_1 and m_2. The new codeword is then modulated into BPSK symbols and transmitted to the destination D. For simplicity we assume that both sources have the same amount of data to transmit to D. JNFC at relays is performed by the following ways:

4.5.2.1 JNFC with Random Linear Network Coding

In this scheme relays encode each source information using LT codes. Let R_i encodes m_1 and m_2 into $\{x_{i1}(1), x_{i1}(2), \ldots.x_{i1}(n)\ldots\}$ and $\{x_{i2}(1), x_{i2}(2), \ldots.x_{i2}(n)\ldots\}$, respectively. Then the output bits of the two LT encoders generate the new code word $\{y_i = y_i(1), y_i(2), \ldots.y_i(n)\ldots\}$ where $y_i(j)$ is either the bitwise XOR of $x_{i1}(j)$ and $x_{i2}(j)$ or just one of them as shown in Fig. 7(a). This implies that the maximum allowed degree of the symbols of $y_i(j)$ is $2k$. Let y_1 and y_2 are the transmitted code words from R_1 and R_2 respectively to destination and are represented as

$$y_1 = \alpha_{11} m_1 . \mathscr{G}_n^3 \oplus \alpha_{12} m_2 . \mathscr{G}_n^4, \tag{4.5}$$

$$y_2 = \alpha_{21} m_1 . \mathscr{G}_n^5 \oplus \alpha_{22} m_2 . \mathscr{G}_n^6, \tag{4.6}$$

where the network coding coefficients $\alpha_{ij}(i, j = 1, 2)$ are drawn randomly from GF(2) and the generator matrix $\mathscr{G}_n^i(i = 3, 4, 5, 6)$ are assumed to be size $k \times n$. However, the generated network codes in this way do not follow Robust Soliton Distribution in degree.

4.5.2.2 JNFC with Robust Soliton Distribution (JNFC-RSD)

One of the attractive features of LT codes is low complexity decoding which is accomplished using Belief Propagation (BP) algorithm that recovers source information k on average $\mathscr{O}(k.\log k)$ symbol operations. However, the efficiency of BP depends on the statistical properties of encoded symbols degree distribution. To ensure low complexity in decoding, the network node should generate the network coded symbols in such a way so that the structure of LT codes is preserved. In (Puducheri *et al.*, 2007)., Puducheri et al proposed a novel way to construct LT from $M = 2$ sources where relay first encodes the received data of each data source onto an LT-like codewords using Deconvolved Soliton Distribution (DSD) as in degree distribution. Relay then generates new code symbol by selectively XOR-ing each pair of symbols in such a way so that the new code follows RSD in degree and has erasure correcting properties similar to LT codes. They also extended their work for $M = 4$ where relay first encodes each source data onto a LT like code words using the doubly deconvolved soliton distribution and then generate new codeword by selective XOR-ing on the encoded data.

However above mentioned approach (Puducheri *et al.*, 2007) is only applicable in particular network scenarios and can not provide any solution when odd number of sources transmit to a single relay. We propose a simple way to generate LT like network codes that follows RSD in degree. In our approach each relay R_i concatenates bits of m_1 and m_2 as one unified input bit sequence which is then encoded by LT code with RSD. By this way network coding is inherently accomplished along with RSD. The encoding processes are depicted in Figure 4.5(b).

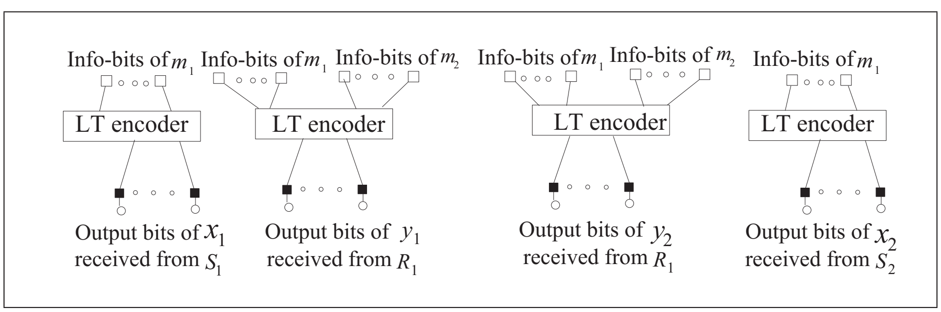


Figure 4.6 The decoding graph at the destination D in JNFC-RSD

4.5.3 Joint Decoding

In fountain encoding the encoded symbols are called output symbols; and the symbols from which these output symbols are calculated are called input symbols. Since we are using (BPSK)

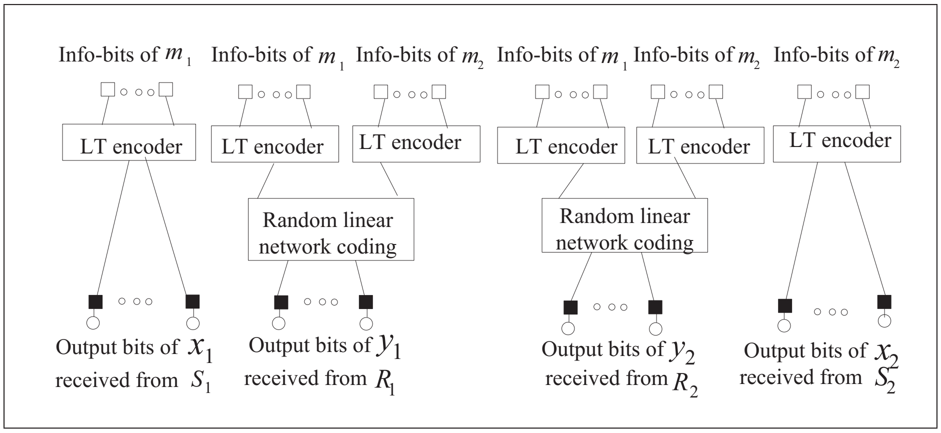


Figure 4.7 The decoding graph at the destination D in JNFC-RLNC

modulation each input and output symbols represent each input and output bits respectively. In type I relay enhanced network destination only receives data from relays where relays generate a long codeword from both sources information and then transmit to destination using Alamouti code. The destination receives the distributed space-time code and performs standard Alamouti decoding first and then LT decoding to retrieve the sources information.

In type II enhanced relay network destination receives packet from both sources and relays. The decoding graph at destination in a type II relay enhanced network are illustrated in Figure 4.6 and Figure 4.7 for JNFC-RSD and JNFC-RLNC, respectively where the circles and rectangles represent output symbols nodes and parity-check nodes of the fountain coding, respectively. The LT graph includes four types of output nodes. The most left part (most right part, respectively) corresponds to the output bits generated by source S_1 (source S_2) and the middle two parts are the output bits generated by relay R_1 and R_2.

Four codewords x_1, x_2, y_1, and y_2 are received at destination. The destination forms a longer code as follows:

$$[x_1\ x_2\ y_1\ y_2] = [u_1\ u_2] \begin{bmatrix} \mathscr{G}_n^1 & 0 & \alpha_{11}\mathscr{G}_n^3 & \alpha_{21}\mathscr{G}_n^5 \\ 0 & \mathscr{G}_n^2 & \alpha_{12}\mathscr{G}_n^4 & \alpha_{22}\mathscr{G}_n^6 \end{bmatrix}. \tag{4.7}$$

The code in Equation 4.7 can be viewed as an integrated code with packets $[u_1\ u_2]$ and generator matrix

$$\mathscr{G}' = \begin{bmatrix} \mathscr{G}_n^1 & 0 & \alpha_{11}\mathscr{G}_n^3 & \alpha_{21}\mathscr{G}_n^5 \\ 0 & \mathscr{G}_n^2 & \alpha_{12}\mathscr{G}_n^4 & \alpha_{22}\mathscr{G}_n^6 \end{bmatrix}. \tag{4.8}$$

Then, decoding of the rateless code is done based on the generator matrix $\mathscr{G}'$.

4.5.4 Transmission Efficiency

We evaluate the performance of the proposed schemes in terms of transmission efficiency rather than outage probability which is a common measure to evaluate the performance of cooperative communications. We assume the channel capacity of the link corresponding to each node pair is denoted, $C_{jk}, j \in \{S_1, S_2, R_1, R_2\}$ and $k \in \{R_1, R_2, D\}$. Since BPSK is used as modulation, the data rate, C_{jk} of a link, j-k can be calculated by (Zhang & Zhang, 2013):

$$C_{jk}^{BPSK} = 1 - \frac{1}{2\sqrt{2\pi\gamma_{jk}}} \int_{-\infty}^{\infty} log_2(1+e^{-x}) e^{-\frac{x-2\gamma_{jk}}{2\gamma_{jk}}} dx, \tag{4.9}$$

where $\gamma_{jk} = |H_{jk}|^2 P_j/\sigma_k^2$. Assuming capacity achieving codes are deploy at each node, then relay R_j will take $N_j^{S_i} = \frac{M_i}{C_{S_i R_j}}$, channel uses, $i = 1, 2$, for successful decoding at the relay R_j. In type I relay enhanced network D only receives information form relays. Therefore in type I network, the total number of channels observed by D , i.e., N_{T_I} to correctly retrieve m_1 and m_2, can be derived from the following equation

$$(N_{T_I} - N_R)C_{RD} = M_1 + M_2, \tag{4.10}$$

where,$C_{RD} =$ $\log_2$det $(I + Es/\sigma^2 HH^H)$ and N_R is the maximum number of channel uses by relays to decode both source messages correctly. H denote the channel matrix of the compound channel from the combined antennas of relay R_1 and R_2 to the antenna of D , which is a 2×1 and $H = [h_{R_1D}\sqrt(G_{R_1D})\ h_{R_2D}\sqrt(G_{R_2D})]^T$. The transmission efficiency is $\zeta_{T_I} = M_1 + M_2/N_{T_I}$ bits/channels.

In type II relay enhanced network, both sources transmit to destination and relays and cease their transmission after their information are successfully decoded at both relay nodes. Therefore to correctly retrieve m_1 and m_2, the total number of channels observed by D , i.e., $N_{T_{II}}$ can be derived from the following equation

$$N_1 C_{S_1D} + N_2 C_{S_2D} + \sum_{j=1}^{2} (N_{T_{II}} - N_{R_j}) C_{R_jD} = M_1 + M_2, \tag{4.11}$$

where $N_1 = max(N_1^{S_1}, N_2^{S_1})$, $N_2 = max(N_1^{S_2}, N_2^{S_2})$ and $N_{R_j} = (N_j^{S_1} + N_j^{S_2})$. Hence the transmission efficiency is $\zeta_{T_{II}} = M_1 + M_2/N_{T_{II}}$ bits/channels.

4.6 Simulation Results

In this section, we conduct simulations to investigate the performance of different JNFC schemes in Rayleigh fading channel. Two sources S_1 and S_2 generate original packets m_1 and m_2, respectively where each of length $k = 500$ bits. Using LT codes then the original packets m_1 and m_2 are encoded into x_1 and x_2, respectively and transmitted to the destination. In the simulation, we consider the scenario where $d_{S_1,R_1} = 4$, $d_{S_1,R_2} = 6$, $d_{S_2,R_1} = 6$, $d_{S_2,R_2} = 4$, $d_{S_1,D} = 10$, and $d_{S_2,D} = 10$. For simplicity, the gaussian noise at each node is assumed to be of the same variance, σ_0^2 and each node has similar transmit power P_0.

4.6.1 Performance Evaluation in Type I Relay Enhanced Network

In type I relay enhanced network, source nodes transmits their messages to relays. The two source nodes stop their transmission when their information block m_1 and m_2 are successfully decoded at relays R_1 and R_2. After successfully decoding the source messages relays concatenate the messages and perform LT coding on the unified message. Relays then synchronize with each other and transmit the encoded symbols using STBC code to eNB until both blocks are successfully decoded at the destination D. We call it JNFC with STBC (JNFC-STBC) and compare it with direct transmissions with relays (DTR), where the source S_i transmits its information block to relay R_i, and after correctly decoding the message R_i performs fountain coding and transmit to the eNB, $i = 1,2$. Here destination does not receive any information in phase I. Let n_1 denote the number of bits received at the each relay before they successfully decode both source messages. D starts decoding after receiving n bits.

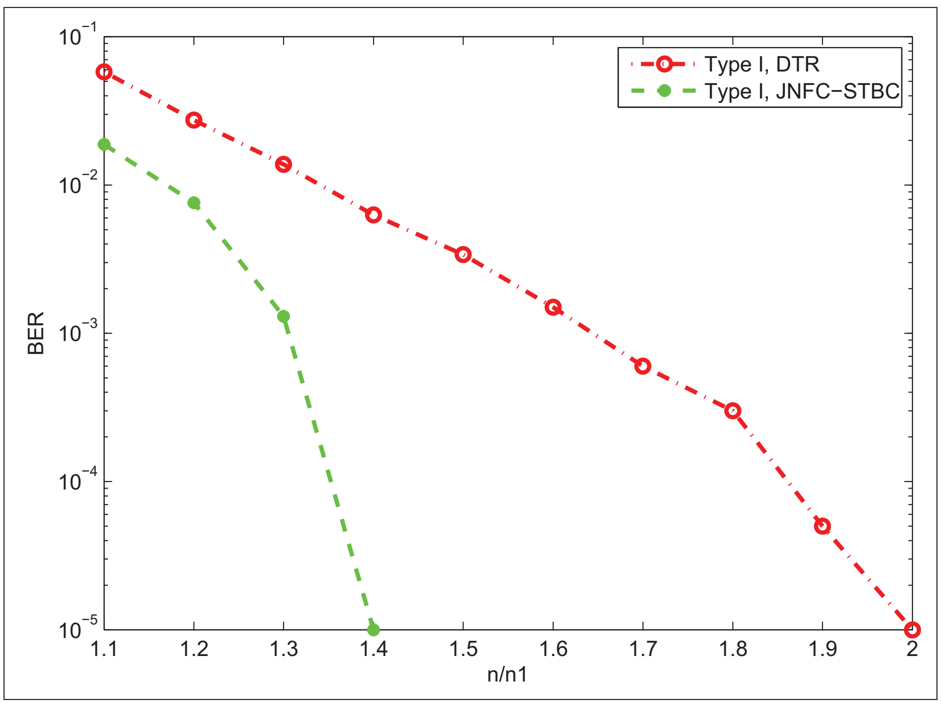


Figure 4.8 Type I enhanced network: bit error rate (BER) Vs n/n_1

Figure 4.8 illustrates the performance of JNFC-STBC and DTR in terms of bit error rate (BER) as a function of n/n_1 ratio. The transmit SNR, i.e., P_0/σ_0^2 is fixed to be 10dB. It is observed that BER of each scheme decreases along with increasing of n/n_1 ratio. Moreover it is observed that JNFC-STBC outperforms DTR. This is because JNFC-STBC gain additional diversity due to the use of STBC with the cost of more signaling overhead during the phase synchronization between two relays. As shown in Figure 4.8 at $n/n1 = 1.4$, JNFC-STBC achieves a BER under 10^{-5} while DTR achieves a BER above 10^{-3}.

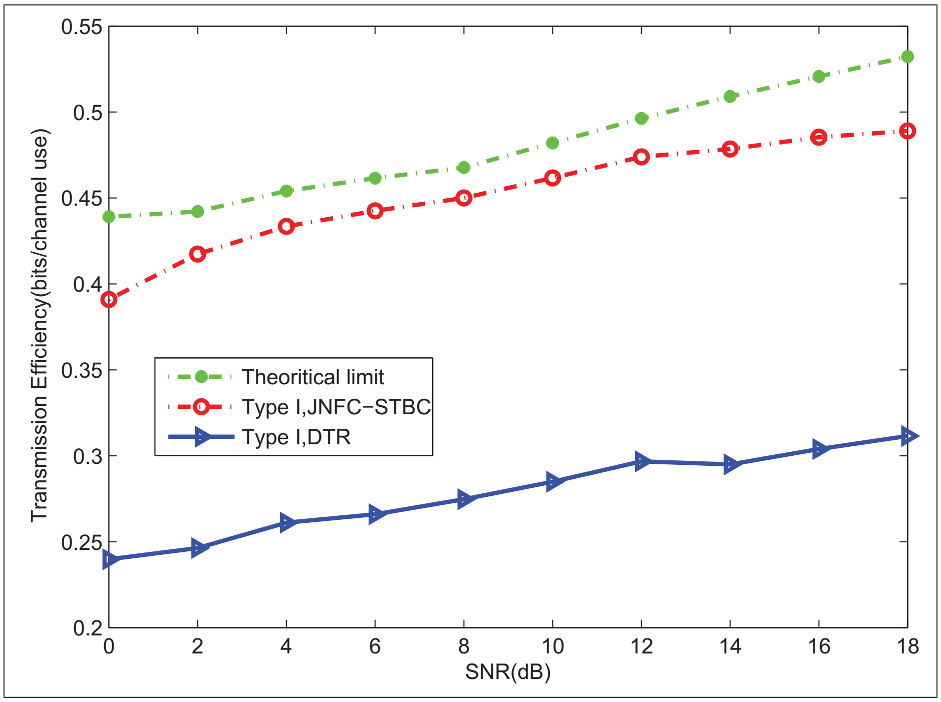


Figure 4.9 Type I enhanced network: transmission efficiency Vs transmit SNR (dB)

Figure 4.9 compares the performance of JNFC-STBC with DTR in terms of transmission efficiency under various transmit SNRs. The bit error rate 10^{-6} is considered as the threshold of successful decoding. It is shown that JNFC-STBC improves the transmission efficiency more than 41% than DTR at transmit SNR=10dB. This is because JNFC-STBC gain additional diversity due to the use of STBC with the cost of more signaling overhead during the phase synchronization between two relays. Since JNFC-STBC improves the performance significantly it may be worth to use STBC in spite of the extra overhead. The theoretical limit of the transmission efficiency is also shown in Figure 4.9. The performance gap between the theoretical limit of the transmission efficiency and the achievable transmission efficiency using JNFC-STBC at transmit SNR=10dB is 8.1345%.

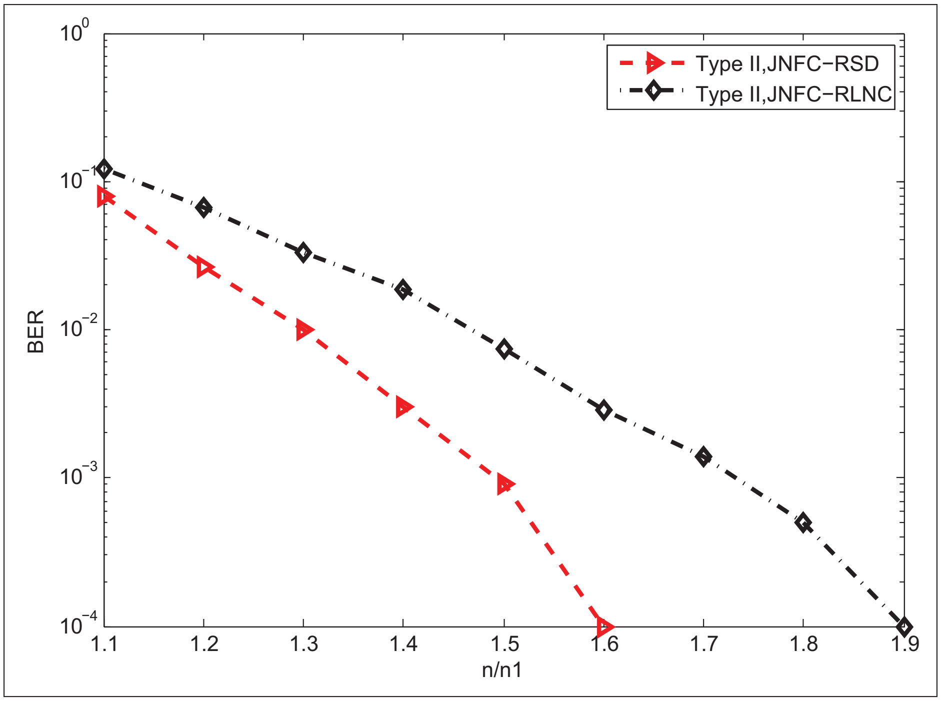


Figure 4.10 Type II enhanced network: bit error rate (BER) Vs n/n_1

4.6.2 Performance Evaluation in Type II Enhanced Relay Network

In type II enhanced relay network, the source nodes broadcast until both relays correctly decode their information. So the number of received information from sources and relays at destination is not equal. The destination collects information from source nodes until they cease their transmission. If the destination is not able to decode source messages correctly it receives more information form relays until it successfully decode the both source messages successfully. Let $n_1/2$ is the number of bits received at each relay to decode each m_i correctly. D starts decoding after receiving n bits, where n_1 and $(n-n_1)$ bits are received from sources and relays, respectively.

Figure 4.10 presents the performance of JNFC-RSD and JNFC-RLNC in terms of bit error rate (BER) as a function of n/n_1 ratio. The transmit SNR is fixed to be 10dB. It is observed that JNFC-RSD outperforms JNFC-RLNC. JNFC-RSD achieves a BER under 10^{-4} at $n/n_1 = 1.6$

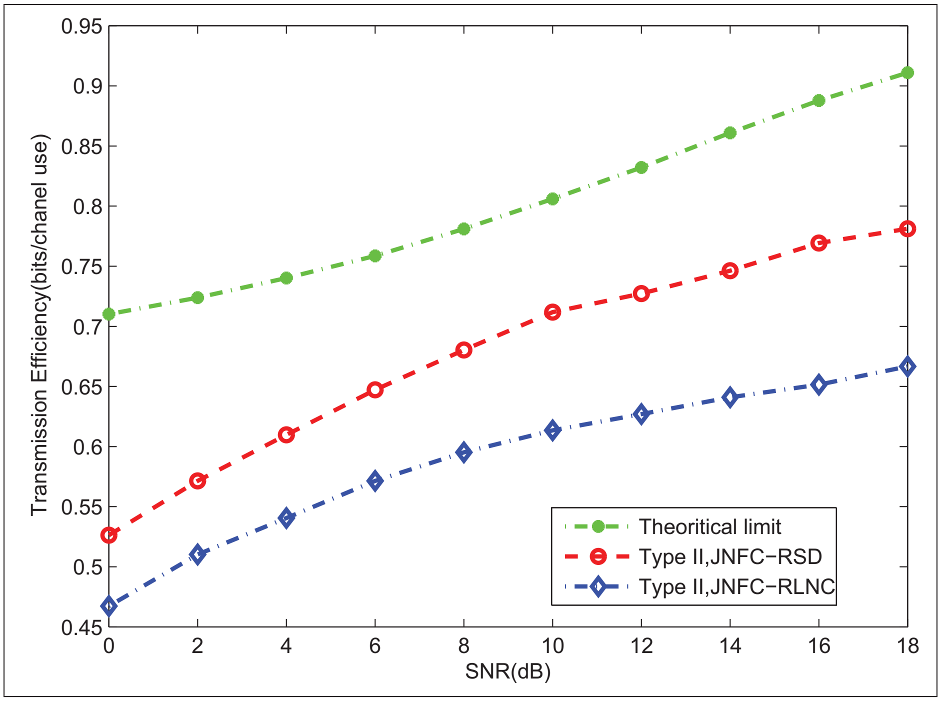


Figure 4.11 Type II enhanced network: transmission efficiency Vs transmit SNR (dB)

while JNFC-RLNC achieves same BER at $n/n_1 = 1.9$. This is because in JNFC-RSD, the relay nodes transmit LT like codewords that follow the RSD in degree distribution.

Figure 4.11 compares the transmission efficiency of proposed JNFC schemes under various transmit SNRs. The bit error rate 10^{-6} is considered as the threshold of successful decoding. We propose two JNFC schemes depending on the coding techniques at relay nodes namely JNFC-RSD and JNFC-RLNC. As shown in Figure 4.11, the JNFC-RSD scheme achieves 14% more efficiency comparing with JNFC-RLNC at transmit SNR=10dB. This is because in JNFC-RSD, relay nodes transmit LT-like codewords that follow the RSD in degree distribution. The theoretical limit of the transmission efficiency is also shown in Figure 4.11. The performance gap between the theoretical limit of the transmission efficiency and the achievable transmission efficiency using JNFC-RSD at transmit SNR=10dB is 14.23%.

4.7 Conclusion

Reliable machine to machine communications is getting more attention as an important IoT performance indicator. LTE is expected to be an important part of the 5G wireless-access solution and the future technology for providing M2M services in IoT paradigm. However, in legacy LTE/LTE-A system Turbo code is used as the forward error correction (FEC) mechanism at the physical layer with hybrid automatic repeat request (HARQ) protocols, where an FEC code in addition to an error detection code is appended to the transmitted packet. In this scheme, a negative acknowledgement (NACK) is sent to the transmitter for retransmission when the receiver fails to decode the information correctly. In severe conditions, ARQ protocols require many retransmissions and have high latency, which result in packet failures and waste of radio resources over the network. Moreover, fixed rate code like Turbo code requires sophisticated channel estimation schemes in order to obtain channel state information (CSI) to determine code rate for adaptation that limits its application to use in many machine type communications. As a possible evolution of LTE-A, in this chapter we presented a set of joint network fountain coding schemes for reliable uplink transmission between M2M devices and eNB. The proposed JNFC seamlessly combines fountain and network coding techniques and thus makes use of the redundancy efficiently. We consider both Type I and Type II enhanced Depending on the coding schemes at relays we proposed JNFC-RLNC and JNFC-RSD. Simulation results demonstrate that in type I enhanced network combined JNFC and space time block code(STBC) outperforms fountain code based direct transmission with relays (DTR) and type II enhance network JNFC with Robust Soliton distribution (JNFC-RSD) outperforms JNFC with random linear network coding(JNFC-RLNC) in terms of bit error rate and transmission.

CHAPTER 5

CONCLUSION AND FUTURE WORK

5.1 Conclusion

Machine-to-machine (M2M) communications over wireless cellular network play a key role in enabling the Internet of Things (IoT) vision by providing ubiquitous connectivity among numerous devices. However, the particular features of M2M communications pose significant challenges over cellular network. It is expected that 5G will be a key enabler for IoT by providing a platform to connect a massive number of devices with stringent energy and transmission constraints. Currently, significant amount of research is ongoing for handling the increased data rate, reduce end-to-end latency, and improve coverage requirements for 5G cellular systems. The overall 5G wireless-access solution will consist of different components, including the evolution of long term evolution (LTE) as well as new technology. The current research and development efforts target to enhance LTE in way so that it can accommodate a large volume of devices and meet their application specific requirements without sacrificing the quality of human to human (H2H) communications.

In this vain in this thesis we developed a framework for reliable data transmission over LTE-A connected heterogeneous M2M network. To accommodate a large volume of devices and reliable data transmission over LTE-A connected M2M network we consider a cluster based architecture where the MTC devices are grouped into number of clusters and traffics are forwarded through some special nodes called cluster heads (CHs) to the LTE-A eNB using single or multi-hop transmissions. We identify CH-to-CH and CH-eNB channels are two main bottlenecks to provide reliable communication in this network and as a solution we employ cooperative communications with fountain coding and network coding to enhance the link robustness.

In Chapter 2, we have focused on reliable intercluster communications and employ cooperative communications and fountain codes to facilitate the data communications between two neighboring clusters. We designed a rateless-coded-incremental-relay selection (RCIRS) algorithm based on greedy techniques to guarantee the QoS requirements with a minimum cost. We also developed two fountain coded cooperative communication protocols namely source-feedback based and non-source-feedback based protocols and studied their performances with different relay selection methods. We have evaluated the performance of our proposed schemes through extensive MATLAB simulations. Our simulation results demonstrated that proposed RCIRS employ less number number of relay nodes than source relay channel based relay selection (SRCRS) to meet the user QoS requirements. It also demonstrated the effectiveness of our proposed protocols than that of non cooperative direct transmission. Furthermore, we observed from simulation results that the source-feedback based protocol always outperforms the non-source-feedback based protocol in terms of transmission efficiency regardless of relay selection method.

In Chapter 3, to overcome the bandwidth bottleneck and enhance the link robustness, we have proposed several joint network and fountain coding (JNFC) schemes that couple channel coding and network coding simultaneously in the physical layer and exploit the redundancy of both codes efficiently to recover erroneous information . Depending on the coding techniques at relay nodes, we proposed two JNFC schemes namely, JNFC with modified LT code (JNFC-MLT) and JNFC with random linear network coding (JNFC-RLNC). We compared the performance of proposed JNFC schemes with separated network and fountain coding (SNFC) where the redundancy provided by network coding is only useful if channel coding is succeeded. Simulation results demonstrated the effectiveness of the proposed JNFC schemes over the direct transmission and SNFC. In addition simulation results demonstrated that JNFC-MLT always outperforms JNFC-RLNC in all metrics.

In Chapter 4, we have studied the uplink LTE HARQ mechanism and found out how it is unsuitable for real time MTC. In legacy LTE network Tubo code is adopted as a channel coding scheme that is a fixed rate code and requires sophisticated channel estimation schemes in order to obtain channel state information (CSI) and to provide channel quality indicator (CQI) feedback for the adaptation mechanisms. In the presence of large amount of devices in the network, the transport of CQI feedback may result in a significant amount of control signaling overhead. Moreover, CQI could be unavailable or outdated at the eNB due to the power saving mode of the M2M devices. Due to the inherit characteristics of fountain code, it can be a potential channel code for uplink transmission between massive MTC devices and LTE eNB. In Chapter 4, we have proposed a frame work for reliable uplink transmission between CHs and base station. The base station could be a LTE-A eNodeB (eNB)or remote radio head of cloud radio access network (CRAN RRH). We consider type I And type II relay enhanced network and propose a set of joint network fountain coding schemes for reliable uplink transmission between M2M devices and eNB. Numerical results are presented to demonstrate the performance of the proposed schemes over type I and type II relay enhanced LTE-A networks. It shows that in type I relay enhanced networks, JNFC with STBC (JNFC-STBC) outperforms the direct transmission with relays (DTR) and in type II enhanced networks JNFC with RSD (JNFC-RSD) outperforms JNFC with RLNC (JNFC-RLNC) in terms of bit error rate and transmission efficiency.

5.2 Future work

a. **Design joint clustering and relay selection**: In this thesis, we developed fountain coded cooperative protocols to facilitate the intercluster communications. However, we did not focus on cluster formation. We proposed relay selection algorithm that selects minimum number of relay nodes to meet the QoS requirements among the homogenous nodes (i.e., they have the same transmission range, processing capabilities, etc). The possible future work includes joint clustering and relay selection in homogeneous and heterogeneous M2M device domain to reduce the signalling overhead. Communication capability, communication link quality, storage status, mobility and and battery life time of each nodes will be considered during clustering and relay selection. Moreover we prefer to select primary and back up cluster heads so that back up cluster heads can play the CH role in the absent of primary cluster head.

b. **Study large scale LTE networks**: We have developed joint network and fountain coding (JNFC) schemes for a 2-2-1 network and used LT code as a channel code. A possible future work is to evaluate JNFC schemes with larger network using Raptor code and evaluate the performance in terms of average delivery delay, bandwidth efficiency and throughput performance.

c. **Joint network and fountain code based HARQ model for NB-IoT:** Narrowband Internet of Things (NB-IoT) is the new frontier in M2M communications, introduced in 3GPP Release 13 for providing wide-area coverage for the Internet of Things (IoT). To enable low-complexity UE implementation, NB-IoT allows only one HARQ process in both downlink and uplink. Since only one HARQ process is allowed we believe fountain code be a potential channel coding technique for reliable data delivery in NB- IoT. The possible future work could be a comprehensive study on improving the efficiency of HARQ scheme in NB-IoT system based on rateless code by utilizing coding and diversity gains.

BIBLIOGRAPHY

3GPP-R2-102296. (2010). *RACH intensity of time controlled devices.* Consulted at http://www.3gpp.org/DynaReport/TDocExMtg--R2-69b--28031.htm.

3GPP-TR-33.812. (2010). *Feasibility study on remote management of USIM application on M2M equipment (Release 9)* (Report n°36.212 V11.2.0). Consulted at http://www.3gpp.org/dynareport/36212.htm.

3GPP-TR-36.300. (2012). *Evolved universal terrestrial radio access (E-UTRA) and evolved universal terrestrial radio access network (EUTRAN), Overall Description* (Report n°36.300 V11.2.0). Consulted at http://www.3gpp.org/dynareport/36-series.htm.

3GPP-TS-36.212. (2010). *Group Radio Access Network; Evolved Universal Terrestrial Radio Access (E-UTRA); Multiplexing and channel coding (Release 9)* (Report n°36.212 V11.2.0). Consulted at http://www.3gpp.org/dynareport/36212.htm.

3GPP.23.368. (2011). *Service requirements for machine-type communication.* (Report n°23.368).

3GPP.23.401. (2011). *Evolved Universal Terrestrial Radio Access Network (E-UTRAN) access.* (Report n°23.401).

3GPP.23.682. (2012). *Architecture enhancements to facilitate communications with packet data networks and applications.* (Report n°23.682).

3GPP.23.887. (2013). *Study on Machine-Type Communications (MTC)and Other Mobile Data Applications Communications Enhancements.* (Report n°23.887). Consulted at http://www.tech-invite.com/3m23/tinv-3gpp-23-887.html.

3GPP.23.888. (2011). *System improvements for machine-type communications.* (Report n°23.888).

3GPP.24.301. (2011). *Non-Access-Stratum (NAS) protocol for Evolved Packet System (EPS) Stage 3.* (Report n°24.301).

3GPP.36.306. (2015). *Evolved Universal Terrestrial Radio Access (E-UTRA), User Equipment (UE) Radio Access Capabilities.* (Report n°36.306).

3GPP.36.321. (2010). *Medium Access Control (MAC) protocol specification* (Report n°36.321).

3GPP.36.322. (2010). *Radio Link Control (RLC) protocol specification* (Report n°36.322).

3GPP.36.323. (2009). *Packet Data Convergence Protocol (PDCP) specification* (Report n°36.323).

3GPP.36.331. (2010). *Radio Resource Control (RRC* (Report n°36.331).

3GPP.36.814. (2010). *Evolved Universal Terrestrial Radio Access (E-UTRA); Further advancements for E-UTRA physical layer aspects* (Report n°36.814).

3GPP.36.888. (2013). *Study on provision of low-cost Machine-Type Communications (MTC) User Equipments (UEs) based on LTE.* (Report n°36.888).

3GPP.37.868. (2013). *RAN Improvements for Machine-type Communications.* (Report n°37.868).

3GPP.37.869. (2013). *Study on Enhancements to Machine-Type Communications (MTC) and other Mobile Data Applications.* (Report n°37.869).

Abdalla, I. & Venkatesan, S. (2013, Aug). A qoe preserving m2m-aware hybrid scheduler for lte uplink. *2013 international conference on selected topics in mobile and wireless networking (mownet)*, pp. 127-132. doi: 10.1109/MoWNet.2013.6613808.

Abouei, J., Dehkordy, S. F., Plataniotis, K. N. & Pasupathy, S. (2011, Sept). Raptor codes in wireless body area networks. *2011 ieee 22nd international symposium on personal, indoor and mobile radio communications*, pp. 2143-2147. doi: 10.1109/PIMRC.2011.6139894.

Ahlswede, R., Cai, N., Li, S. Y. R. & Yeung, R. W. (2000). Network information flow. *Ieee transactions on information theory*, 46(4), 1204-1216. doi: 10.1109/18.850663.

Akyildiz, I. F., Gutierrez-Estevez, D. M., Balakrishnan, R. & Chavarria-Reyes, E. (2014). Lte-advanced and the evolution to beyond 4g (b4g) systems. *Physical communication*, 10, 31 - 60.

Alamouti, S. M. (1998). A simple transmit diversity technique for wireless communications. *Ieee journal on selected areas in communications*, 16(8), 1451-1458. doi: 10.1109/49.730453.

Arslan, H., Cheng, J.-F. & Balachandran, K. (2001). Physical layer evolution for gsm/edge. *Global telecommunications conference, 2001. globecom '01. ieee*, 5, 3050-3054 vol.5. doi: 10.1109/GLOCOM.2001.965987.

Badia, L., Rossi, M. & Zorzi, M. (2006). Sr arq packet delay statistics on markov channels in the presence of variable arrival rate. *Ieee transactions on wireless communications*, 5(7), 1639-1644. doi: 10.1109/TWC.2006.1673074.

Bali, Z., Ajib, W. & Boujemaa, H. (2010, April). Distributed relay selection strategy based on source-relay channel. *Telecommunications (ict), 2010 ieee 17th international conference on*, pp. 138-142. doi: 10.1109/ICTEL.2010.5478647.

Bao, X. & Li, J. (2006a, July). An information theoretic analysis for adaptive-network-coded-cooperation (ancc) in wireless relay networks. *2006 ieee international symposium on information theory*, pp. 2719-2723. doi: 10.1109/ISIT.2006.262148.

Bao, X. & Li, J. T. (2011). Generalized adaptive network coded cooperation (gancc): A unified framework for network coding and channel coding. *Ieee transactions on communications*, 59(11), 2934-2938. doi: 10.1109/TCOMM.2011.091411.070335.

Bao, X. & Li, J. (2006b, May). On the outage properties of adaptive network coded cooperation (ancc) in large wireless networks. *2006 ieee international conference on acoustics speech and signal processing proceedings*, 4, IV-IV. doi: 10.1109/ICASSP.2006.1660904.

Bao, X. & Li, J. (2008). Adaptive network coded cooperation (ancc) for wireless relay networks: matching code-on-graph with network-on-graph. *Ieee transactions on wireless communications*, 7(2), 574-583. doi: 10.1109/TWC.2008.060439.

Berger, C. R., Zhou, S., Wen, Y., Willett, P. & Pattipati, K. (2008). Optimizing joint erasure- and error-correction coding for wireless packet transmissions. *Ieee transactions on wireless communications*, 7(11), 4586-4595. doi: 10.1109/T-WC.2008.070581.

Berrou, C. & Glavieux, A. (1996). Near optimum error correcting coding and decoding: turbo-codes. *Ieee transactions on communications*, 44(10), 1261-1271. doi: 10.1109/26.539767.

Berrou, C., Glavieux, A. & Thitimajshima, P. (1993, May). Near shannon limit error-correcting coding and decoding: Turbo-codes. 1. *Communications, 1993. icc '93 geneva. technical program, conference record, ieee international conference on*, 2, 1064-1070 vol.2. doi: 10.1109/ICC.1993.397441.

Blake, I. F. (2005). Error control coding (s. lin and d. j. costello; 2004) [book review]. *Ieee transactions on information theory*, 51(4), 1616-1617. doi: 10.1109/TIT.2005.844056.

Bletsas, A., Khisti, A., Reed, D. P. & Lippman, A. (2006). A simple cooperative diversity method based on network path selection. *Ieee journal on selected areas in communications*, 24(3), 659-672. doi: 10.1109/JSAC.2005.862417.

Byers, J. W., Luby, M. & Mitzenmacher, M. (2002). A digital fountain approach to asynchronous reliable multicast. *Ieee journal on selected areas in communications*, 20(8), 1528-1540. doi: 10.1109/JSAC.2002.803996.

Byers, J., Luby, M., Mitzenmacher, M. & Rege, A. (1998/10/). A digital fountain approach to reliable distribution of bulk data. 28(4), 56 - 67.

Cabrera, E. (2015). *Addaptive harq (a-harq) for ultra-reliable communications in 5g*. (Master of computer science, Australia, Macquarie University).

Cao, J., Ma, M. & Li, H. (2012, Dec). A group-based authentication and key agreement for mtc in lte networks. *Global communications conference (globecom), 2012 ieee*, pp. 1017-1022. doi: 10.1109/GLOCOM.2012.6503246.

Castura, J. & Mao, Y. (2007). Rateless coding and relay networks. *Ieee signal processing magazine*, 24(5), 27-35. doi: 10.1109/MSP.2007.904814.

Chakrabarti, A., Baynast, A. D., Sabharwal, A. & Aazhang, B. (2007). Low density parity check codes for the relay channel. *Ieee journal on selected areas in communications*, 25(2), 280-291. doi: 10.1109/JSAC.2007.070205.

Chang, H.-C., Shung, C. B. & Lee, C.-Y. (2001). A reed-solomon product-code (rs-pc) decoder chip for dvd applications. *Ieee journal of solid-state circuits*, 36(2), 229-238. doi: 10.1109/4.902763.

Chaoub, A. & Ibn-Elhaj, E. (2014, March). Multimedia transmission over cognitive radio networks using decode-and-forward multi-relays and rateless coding. *Fourth international conference on communications and networking, comnet-2014*, pp. 1-5. doi: 10.1109/ComNet.2014.6840925.

Chen, H., Maunder, R. G., Ma, Y., Tafazolli, R. & Hanzo, L. (2015). Hybrid-arq-aided short fountain codes designed for block-fading channels. *Ieee transactions on vehicular technology*, 64(12), 5701-5712. doi: 10.1109/TVT.2015.2388632.

Chen, J., Cao, X., Cheng, P., Xiao, Y. & Sun, Y. (2010). Distributed collaborative control for industrial automation with wireless sensor and actuator networks. *Ieee transactions on industrial electronics*, 57(12), 4219-4230. doi: 10.1109/TIE.2010.2043038.

Chen, Y., Kishore, S. & Li, J. (2006). Wireless diversity through network coding. pp. 1681 - 6.

Cheng, Z., Castura, J. & Mao, Y. (2009). On the design of raptor codes for binary-input gaussian channels. *Ieee transactions on communications*, 57(11), 3269-3277. doi: 10.1109/TCOMM.2009.11.070095.

Costello, S. L. D. J. (2004). Error control coding : fundamentals and applications (ed. 2nd ed.). Upper Saddle River, N.J. : Pearson-Prentice Hall.

Cui, S., Goldsmith, A. J. & Bahai, A. (2004). Energy-efficiency of mimo and cooperative mimo techniques in sensor networks. *Ieee journal on selected areas in communications*, 22(6), 1089-1098. doi: 10.1109/JSAC.2004.830916.

Darnell, M. (1985). Error control coding: Fundamentals and applications. *Communications, radar and signal processing, iee proceedings f*, 132(1), 68-. doi: 10.1049/ip-f-1:19850011.

Dohler, M., Li, Y., Vucetic, B., Aghvami, A. H., Arndt, M. & Barthel, D. (2006). Performance analysis of distributed space-time block-encoded sensor networks. *Ieee transactions on vehicular technology*, 55(6), 1776-1789. doi: 10.1109/TVT.2006.878729.

Ekici, E., Gu, Y. & Bozdag, D. (2006). Mobility-based communication in wireless sensor networks. *Ieee communications magazine*, 44(7), 56-62. doi: 10.1109/MCOM.2006.1668382.

Etesami, O. & Shokrollahi, A. (2006). Raptor codes on binary memoryless symmetric channels. *Ieee transactions on information theory*, 52(5), 2033-2051. doi: 10.1109/TIT.2006.872855.

Etesami, O., Molkaraie, M. & Shokrollahi, A. (2004, June). Raptor codes on symmetric channels. *Information theory, 2004. isit 2004. proceedings. international symposium on*, pp. 38-. doi: 10.1109/ISIT.2004.1365076.

ExaltedProject. (2011, Aug.). First report on lte-m algorithms and procedures. Consulted at http://www.ict-exalted.eu/fileadmin/documents/EXALTED_WP3_D3.1.pdf.

Fragouli, C., Le Boudec, J.-Y. & Widmer, J. (2006/01/). Network coding: an instant primer. *Computer communication review*, 36(1), 63 - 8.

Ganesan, D., Govindan, R., Shenker, S. & Estrin, D. (2001//). Highly-resilient, energy-efficient multipath routing in wireless sensor networks. pp. 251 - 4. Consulted at http://dx.doi.org/10.1145/501416.501452.

Ghavimi, F. & Chen, H. H. (2015). M2m communications in 3gpp lte/lte-a networks: Architectures, service requirements, challenges, and applications. *Ieee communications surveys tutorials*, 17(2), 525-549. doi: 10.1109/COMST.2014.2361626.

Gong, C., Yue, G. & Wang, X. (2010). Analysis and optimization of a rateless coded joint relay system. *Ieee transactions on wireless communications*, 9(3), 1175-1185. doi: 10.1109/TWC.2010.03.090736.

Guo, Z., Huang, J., Wang, B., Zhou, S., Cui, J. H. & Willett, P. (2012). A practical joint network-channel coding scheme for reliable communication in wireless networks. *Ieee transactions on wireless communications*, 11(6), 2084-2094. doi: 10.1109/TWC.2012.050112.110395.

Guo, Z., Wang, B., Xie, P., Zeng, W. & Cui, J.-H. (2009). Efficient error recovery with network coding in underwater sensor networks. *Ad hoc networks*, 7(4), 791 - 802.

Hall, E. K. & Wilson, S. G. (1998). Design and analysis of turbo codes on rayleigh fading channels. *Ieee journal on selected areas in communications*, 16(2), 160-174. doi: 10.1109/49.661105.

Hassanein, H. & Luo, J. (2006, April). Reliable energy aware routing in wireless sensor networks. *Second ieee workshop on dependability and security in sensor networks and systems*, pp. 54-64. doi: 10.1109/DSSNS.2006.10.

Hausl, C. & Dupraz, P. (2006, Sept). Joint network-channel coding for the multiple-access relay channel. *2006 3rd annual ieee communications society on sensor and ad hoc communications and networks*, 3, 817-822. doi: 10.1109/SAHCN.2006.288566.

Hausl, C. & Hagenauer, J. (2006, June). Iterative network and channel decoding for the two-way relay channel. *2006 ieee international conference on communications*, 4, 1568-1573. doi: 10.1109/ICC.2006.255034.

Hausl, C., Schreckenbach, F., Oikonomidis, I. & Bauch, G. (2005). Iterative network and channel decoding on a tanner graph. 4, 2093 - 2102.

Heinzelman, W. B., Chandrakasan, A. P. & Balakrishnan, H. (2002). An application-specific protocol architecture for wireless microsensor networks. *Ieee transactions on wireless communications*, 1(4), 660-670. doi: 10.1109/TWC.2002.804190.

Ho, T., Medard, M., Koetter, R., Karger, D. R., Effros, M., Shi, J. & Leong, B. (2006). A random linear network coding approach to multicast. *Ieee transactions on information theory*, 52(10), 4413-4430. doi: 10.1109/TIT.2006.881746.

Hou, J., Siegel, P. H. & Milstein, L. B. (2001). Performance analysis and code optimization of low density parity-check codes on rayleigh fading channels. *Ieee journal on selected areas in communications*, 19(5), 924-934. doi: 10.1109/49.924876.

Huq, R. M., Moreno, K. P., Zhu, H., Zhang, J., Ohlsson, O. & Hossain, M. I. (2015, Aug). On the benefits of clustered capillary networks for congestion control in machine type communications over lte. *2015 24th international conference on computer communication and networks (icccn)*, pp. 1-7. doi: 10.1109/ICCCN.2015.7288439.

Hyder, M. M., Khan, R. H. & Mahata, K. (2014, Nov). An enhanced random access mechanism for smart grid m2m communications in wimax networks. *Smart grid communications (smartgridcomm), 2014 ieee international conference on*, pp. 356-361. doi: 10.1109/S-martGridComm.2014.7007672.

ITU-R-M.2083. (2015). Imt vision – framework and overall objectives of the future development of imt for 2020 and beyond. Consulted at https://www.itu.int/pub/R-REP-M.2134-2008.

ITU-R-M.2134. (2008). Requirements related to technical system performance for imt-advanced radio interface(s). Consulted at https://www.itu.int/pub/R-REP-M.2134-2008.

Jung, K.-R., Park, A. & Lee, S. (2010). Machine-type-communication (mtc) device grouping algorithm for congestion avoidance of mtc oriented lte network (pp. 167–178). Berlin, Heidelberg: Springer Berlin Heidelberg. doi: 10.1007/978-3-642-16444-6_22.

Kallel, S. & Haccoun, D. (1990). Generalized type ii hybrid arq scheme using punctured convolutional coding. *Ieee transactions on communications*, 38(11), 1938-1946. doi: 10.1109/26.61474.

Kang, J., Zhou, B., Ding, Z. & Lin, S. (2008, July). Ldpc coding schemes for error control in a multicast network. *2008 ieee international symposium on information theory*, pp. 822-826. doi: 10.1109/ISIT.2008.4595101.

Karalis, A., Joannopoulos, J. & Soljacic, M. (2008). Efficient wireless non-radiative mid-range energy transfer. *Annals of physics*, 323(1), 34 - 48. Consulted at http://dx.doi.org/10.1016/j.aop.2007.04.017. wireless nonradiative energy transfer;mid-range energy transfer;resonant electromagnetic states;slowly-evanescent field patterns;.

Karim, L., Anpalagan, A., Nasser, N., Almhana, J. & Woungang, I. (2014,). Fault tolerant, energy efficient and secure clustering scheme for mobile machine-to-machine communications. *Transactions on emerging telecommunications technologies*, 25(10), 1028 - 44.

Karim, L., Anpalagan, A., Nasser, N., Almhana, J. & Woungang, I. (2014/10/). Fault tolerant, energy efficient and secure clustering scheme for mobile machine-to-machine communications. *Transactions on emerging telecommunications technologies*, 25(10), 1028 - 44.

Kim, J. G. & Krunz, M. M. (2000). Delay analysis of selective repeat arq for a markovian source over a wireless channel. *Ieee transactions on vehicular technology*, 49(5), 1968-1981. doi: 10.1109/25.892598.

Koetter, R. & Medard, M. (2003). An algebraic approach to network coding. *Ieee/acm transactions on networking*, 11(5), 782 - 95.

Kortuem, G., Kawsar, F., Fitton, D. & Sundramoorthy, V. (2010). Smart objects as building blocks for the internet of things. *Internet computing, ieee*, 14(1), 44-51. doi: 10.1109/MIC.2009.143.

Kschischang, F. R., Frey, B. J. & Loeliger, H. A. (2001). Factor graphs and the sum-product algorithm. *Ieee transactions on information theory*, 47(2), 498-519.

Kurs, A., Karalis, A., Moffatt, R., Joannopoulos, J., Fisher, P. & Soljacic, M. (2007). Wireless power transfer via strongly coupled magnetic resonances. *Science*, 317(5834), 83 - 6. Consulted at http://dx.doi.org/10.1126/science.1143254. quantitative model;nonradiative power transfer;self-resonant coils;coupled magnetic resonances;wireless power transfer;.

Laneman, J. N., Tse, D. N. C. & Wornell, G. W. (2004). Cooperative diversity in wireless networks: Efficient protocols and outage behavior. *Ieee transactions on information theory*, 50(12), 3062-3080. doi: 10.1109/TIT.2004.838089.

Le, L. B., Hossain, E. & Zorzi, M. (2007). Queueing analysis for gbn and sr arq protocols under dynamic radio link adaptation with non-zero feedback delay. *Ieee transactions on wireless communications*, 6(9), 3418-3428. doi: 10.1109/TWC.2007.06020038.

Lee, S., Su, W., Batalama, S. & Matyjas, J. D. (2010). Cooperative decode-and-forward arq relaying: Performance analysis and power optimization. *Ieee transactions on wireless communications*, 9(8), 2632-2642. doi: 10.1109/TWC.2010.062310.091554.

Li, Q., Hu, R. Q., Qian, Y. & Wu, G. (2012). Cooperative communications for wireless networks: techniques and applications in lte-advanced systems. *Ieee wireless communications*, 19(2). doi: 10.1109/MWC.2012.6189409.

Li, S., Zhao, S., Wang, X., Zhang, K. & Li, L. (2014). Adaptive and secure load-balancing routing protocol for service-oriented wireless sensor networks. *Ieee systems journal*, 8(3), 858-867. doi: 10.1109/JSYST.2013.2260626.

Liang, W., Xu, K., Tian, H. & Xu, Y. (2011, Nov). Joint design of ldpc and physical-layer network coding for bi-directional relay system in the presence of insufficient timing synchronization. *Wireless communications and signal processing (wcsp), 2011 international conference on*, pp. 1-5. doi: 10.1109/WCSP.2011.6096812.

Lien, S. Y., Liau, T. H., Kao, C. Y. & Chen, K. C. (2012). Cooperative access class barring for machine-to-machine communications. *Ieee transactions on wireless communications*, 11(1), 27-32. doi: 10.1109/TWC.2011.111611.110350.

Liu, X. & Lim, T. J. (2009). Fountain codes over fading relay channels. *Ieee transactions on wireless communications*, 8(6), 3278-3287. doi: 10.1109/TWC.2009.081102.

Loa, K., c. Wu, C., t. Sheu, S., Yuan, Y., Chion, M., Huo, D. & Xu, L. (2010). Imt-advanced relay standards [wimax/lte update]. *Ieee communications magazine*, 48(8), 40-48. doi: 10.1109/MCOM.2010.5534586.

Louie, R. H. Y., Hardjawana, W., Li, Y. & Vucetic, B. (2012, Dec). Distributed multiple-access for wireless communications: Compressed sensing with multiple antennas. *Global communications conference (globecom), 2012 ieee*, pp. 3622-3627. doi: 10.1109/GLOCOM.2012.6503678.

Luby, M. (2002). Lt codes. pp. 271 - 80. Consulted at http://dx.doi.org/10.1109/SFCS.2002.1181950.

Lugand, L. R., Costello, D. J. & Deng, R. H. (1989). Parity retransmission hybrid arq using rate 1/2 convolutional codes on a nonstationary channel. *Ieee transactions on communications*, 37(7), 755-765. doi: 10.1109/26.31168.

Machina-Research. (2015). *Global M2M market to grow to 27 billion devices, generating USD1.6 trillion revenue in 2024*.

Madan, R., Mehta, N. B., Molisch, A. F. & Zhang, J. (2008). Energy-efficient cooperative relaying over fading channels with simple relay selection. *Ieee transactions on wireless communications*, 7(8), 3013-3025. doi: 10.1109/TWC.2008.06090.

Madueño, G. C., Stefanović, . & Popovski, P. (2014a, Dec). Efficient lte access with collision resolution for massive m2m communications. *2014 ieee globecom workshops (gc wkshps)*, pp. 1433-1438. doi: 10.1109/GLOCOMW.2014.7063635.

Madueño, G. C., Stefanović, . & Popovski, P. (2014b). Reliable reporting for massive m2m communications with periodic resource pooling. *Ieee wireless communications letters*, 3(4), 429-432. doi: 10.1109/LWC.2014.2326674.

Malak, D., Dhillon, H. S. & Andrews, J. G. (2016). Optimizing data aggregation for uplink machine-to-machine communication networks. *Ieee transactions on communications*, 64(3), 1274-1290. doi: 10.1109/TCOMM.2016.2517073.

Malkamaki, E. & Leib, H. (2000). Performance of truncated type-ii hybrid arq schemes with noisy feedback over block fading channels. *Ieee transactions on communications*, 48(9), 1477-1487. doi: 10.1109/26.870015.

Mandelbaum, D. (1974). An adaptive-feedback coding scheme using incremental redundancy (corresp.). *Ieee transactions on information theory*, 20(3), 388-389. doi: 10.1109/TIT.1974.1055215.

Marfievici, R., Murphy, A. L., Picco, G. P., Ossi, F. & Cagnacci, F. (2013, Oct). How environmental factors impact outdoor wireless sensor networks: A case study. *2013 ieee 10th international conference on mobile ad-hoc and sensor systems*, pp. 565-573. doi: 10.1109/MASS.2013.13.

Mehta, A., Kagaris, D. & Viswanathan, R. (2006). Throughput performance of an adaptive arq scheme in rayleigh fading channels. *Ieee transactions on wireless communications*, 5(1), 12-15. doi: 10.1109/TWC.2006.1576519.

Mignone, V., Vazquez-Castro, M. A. & Stockhammer, T. (2011). The future of satellite tv: The wide range of applications of the dvb-s2 standard and perspectives. *Proceedings of the ieee*, 99(11), 1905-1921. doi: 10.1109/JPROC.2011.2161848.

Molisch, A. F., Mehta, N. B., Yedidia, J. S. & Zhang, J. (2007). Performance of fountain codes in collaborative relay networks. *Ieee transactions on wireless communications*, 6(11), 4108-4119. doi: 10.1109/TWC.2007.060232.

Nessa, A. & Kadoch, M. (2016). Joint network channel fountain schemes for machine-type communications over lte-advanced. *Ieee internet of things journal*, 3(3), 418-427. doi: 10.1109/JIOT.2015.2497311.

Nessa, A., Kadoch, M., Hu, R. Q. & Rong, B. (2012, Dec). Towards reliable cooperative communications in clustered ad hoc networks. *Global communications conference (globecom), 2012 ieee*, pp. 4090-4095. doi: 10.1109/GLOCOM.2012.6503757.

Nessa, A., Kadoch, M. & Rong, B. (2014a, Feb). Joint network channel fountain scheme for reliable communication in wireless networks. *2014 international conference on* *computing, networking and communications (icnc)*, pp. 206-210. doi: 10.1109/ICCNC.2014.6785332.

Nessa, A., Kadoch, M. & Rong, B. (2016). Fountain coded cooperative communications for lte-a connected heterogeneous m2m network. *Ieee access*, 4, 5280-5292. doi: 10.1109/ACCESS.2016.2601031.

Nessa, A., Kadoch, M. & Rong, B. (2014b). Efficient and reliable communication in wireless relay networks using joint network channel fountain. *JCM*, 9(8), 597–606. doi: 10.12720/jcm.9.8.597-606.

Ngo, H. A., Steven, T., Maunder, R. G. & Hanzo, L. (2010, May). A systematic lt coded arrangement for transmission over correlated shadow fading channels in 802.11 ad-hoc wireless networks. *Vehicular technology conference (vtc 2010-spring), 2010 ieee 71st*, pp. 1-5. doi: 10.1109/VETECS.2010.5494097.

Nguyen, D., Tran, T., Nguyen, T. & Bose, B. (2009). Wireless broadcast using network coding. *Ieee transactions on vehicular technology*, 58(2), 914-925. doi: 10.1109/TVT.2008.927729.

Nikjah, R. & Beaulieu, N. C. (2011). Low complexity selection cooperation techniques using information accumulation in dual-hop relaying networks. *Ieee transactions on wireless communications*, 10(5), 1514-1526. doi: 10.1109/TWC.2011.030311.100589.

Ochiai, H., Mitran, P., Poor, H. V. & Tarokh, V. (2005a). Collaborative beamforming for distributed wireless ad hoc sensor networks. *Ieee transactions on signal processing*, 53(11), 4110-4124. doi: 10.1109/TSP.2005.857028.

Ochiai, H., Mitran, P., Poor, H. V. & Tarokh, V. (2005b). Collaborative beamforming for distributed wireless ad hoc sensor networks. *Ieee transactions on signal processing*, 53(11), 4110-4124. doi: 10.1109/TSP.2005.857028.

Onat, F. A., Adinoyi, A., Fan, Y., Yanikomeroglu, H., Thompson, J. S. & Marsland, I. D. (2008). Threshold selection for snr-based selective digital relaying in cooperative wireless networks. *Ieee transactions on wireless communications*, 7(11), 4226-4237. doi: 10.1109/T-WC.2008.070359.

Osseiran, A., Boccardi, F., Braun, V., Kusume, K., Marsch, P., Maternia, M., Queseth, O., Schellmann, M., Schotten, H., Taoka, H., Tullberg, H., Uusitalo, M. A., Timus, B. & Fallgren, M. (2014). Scenarios for 5g mobile and wireless communications: the vision of the metis project. *Ieee communications magazine*, 52(5), 26-35. doi: 10.1109/MCOM.2014.6815890.

Palanki, R. & Yedidia, J. S. (2004, June). Rateless codes on noisy channels. *Information theory, 2004. isit 2004. proceedings. international symposium on*, pp. 37-. doi: 10.1109/ISIT.2004.1365075.

Park, I., Kim, D. & Har, D. (2015). Mac achieving low latency and energy efficiency in hierarchical m2m networks with clustered nodes. *Ieee sensors journal*, 15(3), 1657-1661. doi: 10.1109/JSEN.2014.2364055.

Peterson, L. L. & Davie, B. S. (2003). Computer networks : a systems approach (ed. 3rd ed).

Puducheri, S., Kliewer, J. & Fuja, T. E. (2007). The design and performance of distributed lt codes. *Ieee transactions on information theory*, 53(10), 3740-3754. doi: 10.1109/TIT.2007.904982.

Qian, C., Chen, H., Ma, Y. & Tafazolli, R. (2013, June). A novel adaptive hybrid-arq protocol for machine-to-machine communications. *2013 ieee 77th vehicular technology conference (vtc spring)*, pp. 1-5. doi: 10.1109/VTCSpring.2013.6691831.

Qiao, D., Gursoy, M. C. & Velipasalar, S. (2009). The impact of qos constraints on the energy efficiency of fixed-rate wireless transmissions. *Ieee transactions on wireless communications*, 8(12), 5957-5969. doi: 10.1109/TWC.2009.12.081600.

Ratnakar, N. & Kramer, G. (2006). The multicast capacity of deterministic relay networks with no interference. *Ieee transactions on information theory*, 52(6), 2425-2432. doi: 10.1109/TIT.2006.874431.

Rico-Alvarino, A., Vajapeyam, M., Xu, H., Wang, X., Blankenship, Y., Bergman, J., Tirronen, T. & Yavuz, E. (2016). An overview of 3gpp enhancements on machine to machine communications. *Ieee communications magazine*, 54(6), 14-21. doi: 10.1109/M-COM.2016.7497761.

Scaglione, A., Goeckel, D. L. & Laneman, J. N. (2006). Cooperative communications in mobile ad hoc networks. *Ieee signal processing magazine*, 23(5), 18-29. doi: 10.1109/MSP.2006.1708409.

Seddik, K. G., Sadek, A. K., Ibrahim, A. S. & Liu, K. J. R. (2008). Design criteria and performance analysis for distributed space-time coding. *Ieee transactions on vehicular technology*, 57(4), 2280-2292. doi: 10.1109/TVT.2007.912145.

Sendonaris, A., Erkip, E. & Aazhang, B. (2003). User cooperation diversity. part i. system description. *Ieee transactions on communications*, 51(11), 1927-1938. doi: 10.1109/T-COMM.2003.818096.

Shah, R. C. & Rabaey, J. M. (2002, Mar). Energy aware routing for low energy ad hoc sensor networks. *Wireless communications and networking conference, 2002. wcnc2002. 2002 ieee*, 1, 350-355 vol.1. doi: 10.1109/WCNC.2002.993520.

Shannon, C. E. (1948). A mathematical theory of communication. *The bell system technical journal*, 27(3), 379-423. doi: 10.1002/j.1538-7305.1948.tb01338.x.

Shariatmadari, H., Ratasuk, R., Iraji, S., Laya, A., Taleb, T., Jäntti, R. & Ghosh, A. (2015). Machine-type communications: current status and future perspectives toward 5g systems. *Ieee communications magazine*, 53(9), 10-17. doi: 10.1109/MCOM.2015.7263367.

Shirvanimoghaddam, M., Li, Y., Dohler, M., Vucetic, B. & Feng, S. (2015). Probabilistic rateless multiple access for machine-to-machine communication. *Ieee transactions on wireless communications*, 14(12), 6815-6826. doi: 10.1109/TWC.2015.2460254.

Shokrollahi, A. (2006). Raptor codes. *Ieee transactions on information theory*, 52(6), 2551-2567. doi: 10.1109/TIT.2006.874390.

Si, P., Yang, J., Chen, S. & Xi, H. (2015). Adaptive massive access management for qos guarantees in m2m communications. *Ieee transactions on vehicular technology*, 64(7), 3152-3166. doi: 10.1109/TVT.2014.2349732.

Soljanin, E., Varnica, N. & Whiting, P. (2006). Punctured vs rateless codes for hybrid arq. pp. 155 - 159.

Song, L., Parhi, K. K., Kuroda, I. & Nishitani, T. (2000). Hardware/software codesign of finite field datapath for low-energy reed-solomon codecs. *Ieee transactions on very large scale integration (vlsi) systems*, 8(2), 160-172. doi: 10.1109/92.831436.

Stanojev, I., Simeone, O., Bar-Ness, Y. & Kim, D. H. (2009). Energy efficiency of non-collaborative and collaborative hybrid-arq protocols. *Ieee transactions on wireless communications*, 8(1), 326-335. doi: 10.1109/T-WC.2009.071403.

Su, W., Sadek, A. K. & Liu, K. J. R. (2005, March). Ser performance analysis and optimum power allocation for decode-and-forward cooperation protocol in wireless networks. *Ieee wireless communications and networking conference, 2005*, 2, 984-989 Vol. 2. doi: 10.1109/WCNC.2005.1424642.

Tacca, M., Monti, P. & Fumagalli, A. (2007). Cooperative and reliable arq protocols for energy harvesting wireless sensor nodes. *Ieee transactions on wireless communications*, 6(7), 2519-2529. doi: 10.1109/TWC.2007.05878.

Tran, T., Nguyen, T., Bose, B. & Gopal, V. (2009). A hybrid network coding technique for single-hop wireless networks. *Ieee journal on selected areas in communications*, 27(5), 685-698. doi: 10.1109/JSAC.2009.090610.

Villa, T., Merz, R., Knopp, R. & Takyar, U. (2012, April). Adaptive modulation and coding with hybrid-arq for latency-constrained networks. *European wireless, 2012. ew. 18th european wireless conference*, pp. 1-8.

Wang, T. & Giannakis, G. B. (2008). Complex field network coding for multiuser cooperative communications. *Ieee journal on selected areas in communications*, 26(3), 561-571. doi: 10.1109/JSAC.2008.4481380.

Wang, X., Liu, Q. & Giannakis, G. B. (2007). Analyzing and optimizing adaptive modulation coding jointly with arq for qos-guaranteed traffic. *Ieee transactions on vehicular technology*, 56(2), 710-720. doi: 10.1109/TVT.2007.891465.

Wang, X., Zhang, Z., Zhang, Y., Zhang, L. & Chen, Y. (2015, Dec). Multi-carrier rateless multiple access: A novel protocol for dynamic massive access. *2015 ieee global communications conference (globecom)*, pp. 1-6. doi: 10.1109/GLOCOM.2015.7417547.

Wilson, M. P., Narayanan, K., Pfister, H. D. & Sprintson, A. (2010). Joint physical layer coding and network coding for bidirectional relaying. *Ieee transactions on information theory*, 56(11), 5641-5654. doi: 10.1109/TIT.2010.2068750.

Winkelbauer, A. & Matz, G. (2012, Nov). Joint network-channel coded multi-way relaying. *2012 conference record of the forty sixth asilomar conference on signals, systems and computers (asilomar)*, pp. 2006-2010. doi: 10.1109/ACSSC.2012.6489392.

Woltering, M., Wubben, D., Dekorsy, A., Braun, V. & Doetsch, U. (2014, May). Link level performance assessment of reliability-based harq schemes in lte. *2014 ieee 79th vehicular technology conference (vtc spring)*, pp. 1-5. doi: 10.1109/VTCSpring.2014.7022973.

Wu, P. & Jindal, N. (2010). Performance of hybrid-arq in block-fading channels: A fixed outage probability analysis. *Ieee transactions on communications*, 58(4), 1129-1141. doi: 10.1109/TCOMM.2010.04.080622.

Xie, Z., Zhang, G., Zhu, H., Bian, D., Jiang, Y. & Hu, J. (2010, Oct). A packet level fec code for deep space communications. *Wireless communications and signal processing (wcsp), 2010 international conference on*, pp. 1-5. doi: 10.1109/WCSP.2010.5633128.

Y. Zhang,R. Yu, M. N. Y. L. S. X. & Gjessing, S. (2012). Cognitive machine-to-machine communications: Visions and potentials for the smart grid. *Ieee network*, 26(3), 6–13.

Yang, F., Ling, S., Xu, H. & Sun, B. (2012, March). Network coding-based aomdv routing in manet. *2012 ieee international conference on information science and technology*, pp. 337-340. doi: 10.1109/ICIST.2012.6221663.

Yang, Q. & Bhargava, V. K. (1993). Delay and coding gain analysis of a truncated type-ii hybrid arq protocol. *Ieee transactions on vehicular technology*, 42(1), 22-32. doi: 10.1109/25.192383.

Yang, S. & Koetter, R. (2007, June). Network coding over a noisy relay : a belief propagation approach. *2007 ieee international symposium on information theory*, pp. 801-804. doi: 10.1109/ISIT.2007.4557108.

Yang, Z. & Host-Madsen, A. (2006, July). Rateless coded cooperation for multiple-access channels in the low power regime. *2006 ieee international symposium on information theory*, pp. 967-971. doi: 10.1109/ISIT.2006.261803.

Yu, M., Li, J. & Blum, R. S. (2007, June). User cooperation through network coding. *2007 ieee international conference on communications*, pp. 4064-4069. doi: 10.1109/ICC.2007.669.

Zhang, J., Zhuang, H., Luo, Z. & Li, Y. (2010, Aug). Opportunistic relay selection for rateless-coded cooperative relaying. *Communications and networking in china (chinacom), 2010 5th international icst conference on*, pp. 1-5.

Zhang, S., Liew, S. C. & Lam, P. P. (2006). Hot topic: Physical-layer network coding. 2006, 358 - 365.

Zhang, Y. & Zhang, Z. (2013). Joint network-channel coding with rateless code over multiple access relay system. *Ieee transactions on wireless communications*, 12(1), 320-332. doi: 10.1109/TWC.2012.120312.120633.

Zhao, Y., Adve, R. & Lim, T. J. (2006, July). Improving amplify-and-forward relay networks: Optimal power allocation versus selection. *2006 ieee international symposium on information theory*, pp. 1234-1238. doi: 10.1109/ISIT.2006.262002.

Zheng, K., Ou, S., Alonso-Zarate, J., Dohler, M., Liu, F. & Zhu, H. (2014). Challenges of massive access in highly dense lte-advanced networks with machine-to-machine communications. *Ieee wireless communications*, 21(3), 12-18. doi: 10.1109/MWC.2014.6845044.

Zhou, Z., Zhou, S., Cui, S. & Cui, J. H. (2008). Energy-efficient cooperative communication in a clustered wireless sensor network. *Ieee transactions on vehicular technology*, 57(6), 3618-3628. doi: 10.1109/TVT.2008.918730.

Zhu, H. & Wang, J. (2009). Chunk-based resource allocation in ofdma systems - part i: chunk allocation. *Ieee transactions on communications*, 57(9), 2734-2744. doi: 10.1109/TCOMM.2009.09.080067.

Zhu, H. & Wang, J. (2012). Chunk-based resource allocation in ofdma systems - part ii: Joint chunk, power and bit allocation. *Ieee transactions on communications*, 60(2), 499-509. doi: 10.1109/TCOMM.2011.112811.110036.

Zhu, Z., Zhang, L. & Wakikawa, R. (2011). Supporting mobility for internet cars. *Ieee communications magazine*, 49(5), 180-186. doi: 10.1109/MCOM.2011.5762816.