Network Coding in Wireless Mesh Networks

By

Umar Ahmad Qureshi

2008-NUST-MS-PhD IT-30

Supervisor

Dr. Anjum Naveed

NUST-SEECS

A thesis submitted in partial fulfillment of the requirements for the degree of Masters of Science in Information Technology (MS IT)-9

In

School of Electrical Engineering and Computer Science,
National University of Sciences and Technology (NUST),
Islamabad, Pakistan.

(April 2011)

Approval

It is certified that the contents and form of the thesis entitled "**Network Coding in Wireless Mesh Networks**" submitted by **Umar Ahmad Qureshi** have been found satisfactory for the requirement of the degree.

Advisor: **Dr. Anjum Naveed**

Signature: ________________

Date: ____________________

Committee Member 1: **Dr. Zawar Hussain Shah**

Signature: ____________________

Date: ____________________

Committee Member 2: **Dr. Khawar Khurshid**

Signature: ____________________

Date: ____________________

Committee Member 3: **Dr. Adnan Khalid Kiani**

Signature: ____________________

Date: ____________________

Abstract

Worldwide Wireless Mesh Networks have been advocated as cost effective and robust solution for providing Internet connectivity for the last mile networks. These networks can utilize MR-MC nodes and use 802.11 WLAN technologies for communication. WMNs utilize Ad hoc routing protocols for their communication. This gives them the ability to perform multi hop routing in wireless settings. However these networks frequently face traffic congestion usually near the gateway nodes where a lot of data is held up in uplink and downlink queues. This data has special properties which can be utilized to help improve the throughput of the network and reduce the queues for nodes near the gateway. In recent years many solutions have been proposed to handle the situations arising in WMNs. For our proposal we advocate the use of Network Coding for capacity improvement for these networks. Network coding can be applied between routing-MAC layers with little or no modifications to the protocols being run at these layers. The concept has been thoroughly examined for single radio single channel settings, and has shown to bring the information capacity of a network closer to its ideal limit. In our study we have shown that the use of network coding improves the utilization of multiple channels in MRMC in WMN and proposed an algorithm to improve the opportunity capturing ability of Netcoding operation.

Certificate of Originality

I hereby declare that this submission is my own work and to the best of my knowledge it contains no materials previously published or written by another person, nor material which to a substantial extent has been accepted for the award of any degree or diploma at National University of Sciences & Technology (NUST) School of Electrical Engineering & Computer Science (SEECS) or at any other educational institute, except where due acknowledgement has been made in the thesis. Any contribution made to the research by others, with whom I have worked at NUST SEECS or elsewhere, is explicitly acknowledged in the thesis.

I also declare that the intellectual content of this thesis is the product of my own work, except for the assistance from others in the project's design and conception or in style, presentation and linguistics which has been acknowledged.

Author Name: **Umar Ahmad Qureshi**

Signature: ________________

Acknowledgment

Firstly , All praise to Almighty Allah for he is both merciful and benevolent, and nothing could have been possible without His blessings.

I would like to thank my parents for their support and encouragement. They always supported and encouraged me to do my best in all matters of life. I dedicate this work to them. I would also like to thank my family for their love and prayers for the successful completion of this work.

My heart felt thanks to my committee members, Dr. Zawar Hussain Shah, Dr. Khawar Khurshid, Dr. Adnan Khalid Kiani, and to everyone at CERN Lab, and all others who contributed in any way towards the successful completion of this thesis.

Finally, this thesis would not have been possible without the expert guidance of my advisor, Dr. Anjum Naveed, who has provided me with limitless support during this research. Despite all the assistance provided by Dr. Anjum Naveed and others, I alone remain responsible for any errors or omissions which may unwittingly remain.

Umar Ahmad Qureshi

Contents

List of Abbreviations

Abbreviations	Descriptions
WMN	Wireless Mesh Network
MR-MC	Multi Radio Multi Channel
Netcoding	Network Coding
AODV	Ad hoc On demand Distance Vector
DSR	Dynamic Source Routing
WiFi	Wireless Fidelity
WLAN	Wireless Local Area Network
WIMAX	Worldwide Interoperability for Microwave Access
DSL	Digital Subscriber Line
WOBAN	Wireless Optical Broadband
HDTV	High Definition Television
XOR	Exclusive Or
GF	Galois Field
noCoCo	Near-Optimal Coordinated Coding
COPE	Coding Opportunistically
TCP	Transmission Control Protocol
MORE	Multi-path Opportunistic Routing Engine
C and M	COPE and MORE
BSF-CA	Breadth First Search Channel Assignment
ARP	Address Resolution Protocol
MAC	Medium Access Control
RTS	Request To Send
CTS	Clear To Send
IP	Internet Protocol
RIP	Routing Information Protocol
OSPF	Open Shortest Path First

List of Figures

List of Tables

Chapter 1

Introduction

For the past few years Wireless Mesh Networks have risen as a promising technology for the deployment of affordable and easily available mobile devices, for the last mile network. These networks are capable of providing connectivity using a number of last mile solutions. The most notable of these solutions are the 802.11 (WLAN) and 802.16 (WIMAX) technologies. In wireless mesh networks (Fig.1.1) a limited number of nodes are provided Internet connectivity, these nodes are known as Gateway nodes. In these networks most/all of the traffic generated by the nodes is specified for the Gateway nodes. These Internet gateways are the main providers of services for these networks, providing access to services such as VoIP, ftp and http. Wireless Mesh Networks (WMN) although capable of incorporating both WLAN and WIMAX technologies are usually deployed utilizing the 802.11 (a/b/g) WLAN devices. The choice for this technology is mainly because of the market penetration i.e. cost effectiveness, ease of availability, as well as ease of maintenance and decentralized configurations using Ad hoc mode. The 802.11 WLAN devices for WMNs utilize Ad hoc routing protocols (AODV [17], DSR [18]) for communication within a network. This makes the deployment and fault tolerance of such networks very cost effective and flexible.

WMNs are different from pure ad hoc networks in a number of ways. Although none of these demarcations are strictly followed the general characteristics of WMNs are

- In WMN the communicating nodes, although capable of mobility, show limited or no mobility.

- The nodes in WMNs normally are not short of power i.e. they are not battery operated.

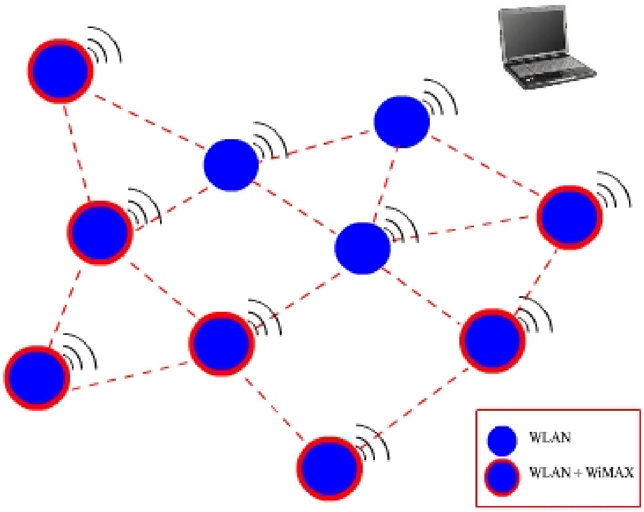

Figure 1.1: Wireless Mesh Network

- These nodes also have considerable processing powers.

The use of Ad hoc networking protocols helps to provide these nodes some intelligence, thereby reducing the maintenance overhead. By deploying Ad hoc routing protocols under the Ad hoc wireless [19] mode specified for 802.11 radios the devices become capable of finding their path towards the Internet Gateway and from then on to provide desired services. While providing cheap and flexible Internet access, the WMN over time has come to face a number of issues that hinder its performance. The most notable and severe of them is the congestion at gateway links. As the number of devices utilizing the network increase the traffic starts to gather at the nodes that are frequently accessed i.e. near the gateway. At these links queues begin to form containing packets that need to be transferred to/from the gateway node. In recent years the challenges related to WMN have seen several different solutions addressing the problem through different dimensions such as increasing efficiency in route selection/path formation, improving channel assignment/utilization and improving the channel capacity through physical technologies. A major advancement in this regard is the introduction of multiple radio interfaces that are capable of tuning to different radio channels. Known as the Multi Radio- Multi Channel technologies (MR-MC), these devices have increased the available bandwidth of wireless nodes by several folds. The nodes now are able to have 2 or 3 radios, each of which is an implementation of the 802.11 standard. Thus with the introduction of

MR-MC nodes the issue of efficient channels assignment also rose, as it was now possible for a transmitter and receiver present between two nodes to be out of sync. Such a scenario would of course lead to a break down in network topology and unnecessary partitioning of the network. Initial proposals in this regard [14],[15] suggested the use of a single default channel to be present at each node dedicated to serving of control traffic and to perform configurations for the other radios. Even though these suggestions provided highly reliable networking options, the obvious disadvantage of dedicating a whole radio solely for the purpose of control packet exchange resulted in dismissal of these requests as practical. We will in our project try to address this issue and will make arguments in this regard. Network Coding was another solution to be considered for improving the efficiency of a wireless network. Network coding was known to have shown improved throughput over satellite links and also for peer 2 peer networks. Having its roots in information theory and coding theory, network coding has the ability to address the information aspect of a computer network. For reducing redundant transfers of information over a network, network coding is a strong candidate while considering the solutions for the problems faced by WMNs. The rest of the chapter contains detailed explanation of Wireless Mesh Networks in section 1.1. Section 1.2 discusses the issue of congestion in WMN. Section 1.3 gives a brief description about network coding and its operation while 1.4 discusses the open issues regarding WMN.

1.1 Wireless Mesh Networks

Wireless Mesh Networks, also known as WMNs, are networks consisting of decentralized nodes. These nodes are capable of self configuration and maintenance because of the use of intelligent routing protocols such as AODV [17] and DSR [18]. These networks consist of two types of devices

- Mesh Routers
- Mesh Clients

In Fig.1.2 Mesh Routers in addition to having ad hoc routing protocols deployed on them also have limited mobility and significant processing capabilities. Mesh clients on the other hand have limited resources and can have advanced mobility.

Mesh routers are connected to each other via wireless links as well. These links are maintained by ad hoc routing protocols and are capable of fault tolerance. The links that constitute the path for Mesh Routers are known as

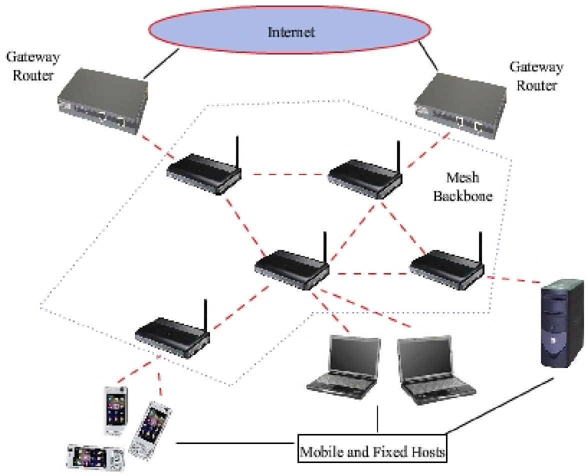

Figure 1.2: WMN Deployment strategy

Mesh Backbone. Some of the mesh routers are assigned additional capabilities and are to serve as a bridge for other technologies such as WIMAX and DSL. These routers are known as Gateway routers. The rest of the mesh routers are used to extend the capabilities of the Gateway routers. The reason for having fewer gateways is usually financial constraints, as dense deployment of WIMAX/DSL nodes is not always a cost effective solution. By having multi hop communication between WMN nodes several devices are capable of sharing an expensive resource link. The sharing of expensive resource links is currently the most common use for WMNs, having been deployed in over 30 cities in USA both for municipal and public use. Another use of WMN is to provide network coverage in disaster struck or remotely located areas. As the WMN devices are intelligent and capable of self configuration depending upon the environment, these networks are extremely effective in these situations. Also temporary venues e.g. concerts and hotels and resorts have also their use beneficial. Another related use of WMNs is the application of WOBAN (Wireless Optical Broadband). These networks consist of wireless nodes usually of 802.11 technologies and the gateway routers in this case are equipped with optical fibre interfaces. These networks are provided to dense urban environments for providing cost effective deployments for applications like HDTV where high bandwidth connections are desired. Specifically, 802.11n standard with approximate bandwidth of 400Mbps (shared) is designed for these situations.

1.2 Congestion in WMNs

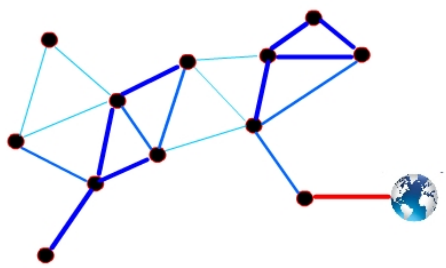

Figure 1.3: Congestion in WMN

In the previous section we have explained in detail how a WMN network is organized and operated. From the Fig.1.3 it is observed that when a WMN node decides to send data towards its required destination it passes it to its immediate neighbor for routing. The neighbouring node then resends this packet further towards its defined next hop. However in WMN all nodes are capable of not only routing a packet towards its destination but of also generating their own data. This means that data to be transmitted at any given node will be a cumulative load of all the previous nodes that need to reach the same destination. On the other hand, data will arrive at the gateway from the Internet for all the nodes that have requested it. Now both gateway node and its neighbouring nodes are at a fierce competition to gain access to the channel and if channel capacity is reached by these nodes then queues will start to form at these nodes. Similarly if the nodes in a WMN are running such that the communication held is in the form of Peer-to-peer, with in the network, communication then the chances of reaching congestion are reduced. However congestion can still occur if nodes in one section of the network decide to communicate with nodes of another section of the network, and there exist very few nodes in-between these two sections. Now as the communication is peer-to-peer the nodes in the reciprocating section will also choose these same nodes for reply, thereby causing choking of the connecting links and formation of queues as a result. The above mentioned scenarios occur very frequently in most WMNs and are a major hindrance in achieving the full potential of these networks.

1.3 Network Coding

Network coding was first introduced, by Yeung et al. [20], in landmark paper as means of reducing latency and achieving increased efficiency for satellite links. Network coding was proposed as means to replace the existing store-&-forward routing technique. The technique aims to target the information aspect of the data being forwarded over the network and criticizes the treatment of data packets as a commodity flow. Traditional store-and-forward routing considers data only as a commodity flow. But according to the theory of information flow "information content" that is present at any non source node can be derived from the total information that is received from that node. This theory implies that the total information transmission can be achieved through fewer packet transfers between nodes.

1.3.1 Definition

Network coding is a method of optimizing the flow of digital data in a network by transmitting digital evidence about messages. The "digital evidence" is itself, a composite of two or more messages. When the bits of digital evidence arrives at the destination, the transmitted message is deduced rather than directly reassembled. [21] The concept of network coding in WMNs can be

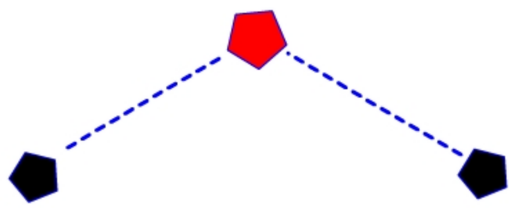

Figure 1.4: Three Node

explained by a simple example Fig.1.4. Suppose we have a 3 node network. The setting for this network goes as

- Node A needs to send data packets to node C
- Node C needs to send data packet to node A
- Both nodes A and C need to traverse node B as an intermediate hop to reach their respective destinations.

Now under normal routing scenario to handle the above situation the process goes as such.

1. To reach node C node A transmits data to node B
2. To reach Node A node C transmits data to node B
3. Node B sends the packet from A towards C
4. Node B sends the packet from C towards A

As can be seen the process requires 4 transactions to be completed before both packets reach their respective destinations. Now under network coding the same scenario is played a little differently.

1. To reach node C node A transmits data to node B and saves it
2. To reach node A node C transmits data to node B and saves it
3. Node B now has two packets from both sources. It therefore performs an XOR operation on both packets to combine both packets into a single encoded signature and then transmits this signature to both nodes simultaneously as a broadcast.

If we take a look now we see that a total number of 3 transactions have taken place. Now the situation at node A is that it has a copy of the packet which it has sent to B and a signature which it has received from B. Node A performs an XOR operation on the packet and the signature. As the signature received is made of the following relation a$\oplus$b, by performing another XOR operation will result in the following relation at A i.e. (a$\oplus$b)$\oplus$a. we will now have packet b as the remainder of the above calculation. Thus packet 'b' has reached destination. These "Exchangeable" packets are a frequent occurrence in WMNs, especially near the gateway nodes. Our aim is to identify these packets and utilize the opportunity they provide in reducing the number of wireless packet transactions for reduction in congestion.

1.4 Open issues

Until now network coding algorithms have been developed and tested over WMNs which possess single radio capable nodes. Algorithms performing and maximizing network coding gain have been developed to utilize single radio deployments and, the gains and losses resulting only from those scenarios have been debated. Also very little work has been done with reference to

Network Coding in MR-MC technologies. Majority of the work done in this regard is for improvement through channel assignment. Most of the solutions proposed, utilize gain maximizing algorithms which were developed for single channel deployments. Our aims are to prove that such assumptions degrade the networks performance and to find out the improved gains through the development of algorithms that are capable of focusing on the multi radio property of the system to improve the overall throughput of the network. The rest of the thesis document comprises of the detailed review of the existing body of work with regards to different aspects of our project in Chapter 2 Literature Review. Chapter 3 contains discussion about our main idea and hypothesis of the project. Chapter 4 ”Implementation” explores the design decisions that were taken and the reasons behind them. And lastly chapter 5 contains the evaluations and analysis of the solution in simulated environments.

Chapter 2

Literature Review

In this chapter we explore the literature presented in relation to the solution. Since the solution takes in to consideration different aspects of the wireless networking paradigm, we have therefore divided this chapter into 3 sections. Section 2.1 discusses the various channel assignment strategies developed for MR-MC traditional routing scenarios. Section 2.2 discusses various channel assignment strategies which have been developed with specific focus to the network coding and its performance improvement. Whereas, in section 2.3 we discuss the major network coding implementing algorithms and the issues regarding their performance.

2.1 Channel Assignment

Regarding channel assignment for traditional routing in MR-MC there is rich literature available which addresses the issue. Works in [6] classify channel assignment algorithms in three broad categories, namely:

1. Static Channel Assignment schemes
2. Dynamic Channel Assignment schemes
3. Hybrid Channel Assignment schemes

Static channel assignment schemes tend to be the simplest of the three. These schemes assign channels to available radios in either a predefined fashion or by using a centralized algorithm at the time of network initialization. The assignments performed by these schemes remain constant throughout network's lifetime (or for long periods of time). Since these techniques do not frequently switch channels the performance achieved by these techniques is currently much greater than the ones which do channel switching. This is because these techniques have very low control overhead and no channel switching delays. Works in [9] and [10] propose static channel assignment schemes for MR-MC WMNs. As opposed to this the dynamic channel assignment techniques aim at switching channels on per packet bases or for a group of packets. The most detrimental performance effect in this form of channel assignment is the presence of considerable channel switching delays in currently available radio technologies. Due to this reason these techniques are mostly aimed at future physical implementations of wireless devices which are capable of switching channels with negligible switching delays. These techniques for the time being provide comparatively poor throughput under practical scenarios and are considered mostly for theoretical discussions. In [11], [12], [13] dynamic channel assignment schemes have been proposed which assign channels to radios dynamically for short durations. The third category for channel assignment is the hybrid channel assignment schemes. These schemes fix one radio present at a device to a single predefined channel while performing dynamic switching for the rest of available radios. These schemes aim to take the advantages of both static and dynamic switching schemes while aiming to suppress their shortcomings. The fixed channel used in these schemes is commonly called as the default channel. The default channel is mostly used to handle control traffic, such as path formations, joining requests and hello packets etc, and also broadcast and multicast traffic. Works in [14] and [15] present hybrid channel assignment schemes for MR-MC WMNs. Fig 2.1 shows a typical Hybrid Channel Assignment Scheme.

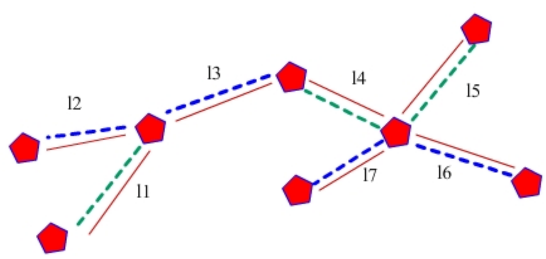

Figure 2.1: Hybrid Channel Assignment Scheme

In [2] the authors have performed XOR operations on overlapping flows to achieve coding gains in access of 50% for their defined scenarios. The solution proposed however, is deployed and analyzed for single channel environment.

Our hypothesis is that if a similar solution is developed and deployed for MR-MC nodes with hybrid or static channel assignment, it will be able to enhance network throughput even further. For the purpose of performing network coding we need to have at least one common channel available at the intermediate links. This channel will be able to carry the encoded packets, generated as broadcast traffic. For this operation either a "static channel assignment scheme" is applied, which will have a common channel available among all nodes, or one of the "hybrid channel assignment schemes" which has support for a default channel common among all (or at least among immediate neighboring) nodes.

2.1.1 Network coding and MR-MC WMNs

Implementations of network coding technology have been developed and applied on several different layers. The main purpose of this technology is to achieve the higher network capacity. The applications of this technology exist for Application, Data link and Physical layers. For each layer number of solutions have been presented which aim to exploit different characteristics present at these layers. Our main concern lies with the implementations available for the data link layer. For this layer the use of coding over GF (2), which allows intermediate relay nodes to opportunistically XOR incoming packets [2], has been found to be both simple and effective. To the best of our knowledge regarding network coding there exists no channel assignment technique for single radio nodes that consider exploring the relation between channel assignment strategies and network coding such that optimal performance can be achieved without interference. Our reason for this is, since the deployed nodes use single radio interfaces and currently they are unable to switch channels at per packet basis effectively, nodes are assigned channels for prolonged periods of time. By allocating channels based on network coding better coding opportunities are availed but this will also end up partitioning the network for all other non encodable transmissions. Since in practical scenarios non code-able transmissions also form a major share of total transmissions, this will adversely affect performance and is therefore avoided. In [8] authors have formulated a joint routing and channel assignment strategy which aims at assigning channels with regard to routing changes in such a way that the interference between them is minimized. For MR-MC WMNs, the effects of channel assignment strategies in relation to network coding have been explored in [1] as well. In this paper authors have discussed that the performance of network coding is maximized if there exists a channel assignment scheme to support the technique. the authors have also proposed a heuristic algorithm regarding channel assignment which tends to provide a common channel where coding opportunity exists and flexibly assigns channels to the rest of available radios. Using this, authors claim to achieve throughput enhancements of up to 21% over the optimized channel assignment scheme discussed in [8]. Another study [7] shows the extended network coding algorithm to perform XOR operation on three packets or more instead of two. Alongside this, a channel assignment strategy is proposed which allocates channels in such a way that neighboring nodes with netcoded transmissions will end up being on the same channel. This way these nodes will be able to overhear each other's transmission and therefore will successfully be able to decode the network coded packet.

2.1.2 Network coding maximizing algorithms

Regarding improvement of network coding opportunities, literature shows an extensive work [2], [4], [5] that tends to exploit different aspects of network traffic dynamics. In [2] the authors present COPE (Coding Opportunistically); a popular algorithm which aims to maximize channel utilization by opportunistically selecting packets for network coding. COPE operates by continuously scanning transmission queues for pair-able packets. If it finds packets which it can combine and encode for transmission it immediately avails the opportunity and performs network coding. The logic behind COPE is not maximization of network coding opportunities but rather maximizations of network throughput. As under practical scenarios queues are formed only if the channel capacity is being utilized to its maximum possible value. However in its eagerness to reduce queues COPE fails to see the bigger picture and instead wastes many coding opportunities as described in [4]. This leads to an increase in network throughput but not to maximization of coding gain/opportunities. Compared to that, the authors of [4] present another coding opportunity maximizing algorithm named noCoCo (Near-Optimal Coordinated Coding). Unlike COPE, noCoCo's aim is to maximize network coding by increasing the opportunities available. noCoCo's operates by holding a packet at a node for as long as its pair packet arrives from other direction. Upon receiving the packet, noCoCo performs network coding and transmits the packet through broadcast. Since noCoCo performs network coding on 1:1 basis, it also requires two opposite and overlapping flows which have to be exactly equal in magnitude. In order to fulfill these criteria noCoCo applies a backpressure technique which effectively tells the greater of the two flows to reduce its transmission rate to match the transmission rate of the smaller flow. This equals both the overlapping flows and supports noCoCo's operation. The application of noCoCo does achieve its aim of maximizing network coding opportunities available within the network.

However this algorithm fails to achieve the broader objective of maximizing network's coding gain. A closer look at the algorithm reveals that the solution effectively tells transmitting nodes to limit their transmission rates. This behavior will limit the overall network throughput as nodes will no longer be able to inject data into the network even though there is still capacity available. Also due to prolonged holding of packets the higher layer TCP recovery mechanisms are likely to be initiated and this will further affect the network performance. In [22] authors propose MORE, Multi-path Opportunistic Routing Engine, a routing protocol which aims to maximize network throughput using network coding. The main objective of MORE is to provide an error free channel which results in lesser packet drops and thus better throughput. The solution uses network coding concepts, by performing coding on several packets present at a node, all of which are to be forwarded to the same destination. The solution applies random linear codes to achieve this. However, in this solution, the number of transmissions from source to destination remains same as traditional routing (under error free assumptions). Authors in [5] have proposed a solution called as C and M (COPE and MORE) algorithm. This algorithm combines both inter-flow (COPE) and intra-flow (MORE) network coding algorithms. The purpose of this combination is to reduce the packet loss rate which adversely affects network coding performance. This reduction in packet loss rate therefore translates to improved coded transmission rates and thus to improvement in network throughput. The research works presented above have one major objective in common, which is they aim only to increase the network coding opportunities whereas, the major objective of our proposed solution is improvement in throughput of the network. In our simulation and analysis we have found that there exist certain instances in a network's dynamics which advocate that instead of availing every opportunity presented for network coding better throughput can be achieved by reducing the amount of network coding. Also in some cases it is beneficial if we avoid network coding completely for a particular path or link. In order to deal with these situations we have developed a network coding maximizing algorithm which is capable of responding to these situations. By turning network coding on/off at different nodes according to requirement we gain more control over the behavior of network traffic. The solutions presented in the above papers are in our view inflexible to react to different situations and cannot provide optimal network coding gain even though they tend to increase network coding opportunities. In order to set up the test bed the use of either static or hybrid channel assignment schemes are suggested. For the simulations the channel assignment strategy BSF-CA discussed in [14] is used. This is a hybrid channel assignment scheme which allocates a fixed single channel to one of the radio present at a node and assigns channels to rest of the nodes based on least level of interference. The default channel for this strategy is used for, preventing the network from partitioning and handling broadcast traffic.

Chapter 3

Netcoding for network capacity improvement

In WMNs the deployment of MRMC nodes have resulted in the development of an array of different channel assignment schemes [9], [10], [11], [12], [13], [14], [15]. These schemes present their claims for several different configurations for assigning channels to WMN radios and utilizing them efficiently. In most of the channel assignment schemes [14], [15] there exists however a default channel that is used to maintain the topology of a network by carrying control traffic. This channel is usually dedicated to control and/or broadcast traffic. As the occurrence of control and broadcast traffic is very low in most networks a lot of channel capacity is left unused and thus wasted. Several schemes have been proposed [11], [12], [13] that aim to counter this problem. In our research we are showing that the use of a default channel is beneficial and can be utilize very effectively through the use of network coding. The section 3.1 of this chapter addresses the channel assignment scheme used in the project. Section 3.2 discusses and explains the behavior of the proposed algorithm. Section 3.3 lists various parts of the algorithm while section 3.4 and 3.5 explain the advantages and performance of the solution. To implement our proposed scheme following two objectives need to be met:

- A suitable Default channel capable channel assignment scheme
- A suitable Network coding opportunity maximizing algorithm

3.1 Channel Assignment Scheme

We know that in order to perform network coding we need a broadcast medium/channel over which the encoded packet will be transmitted. This

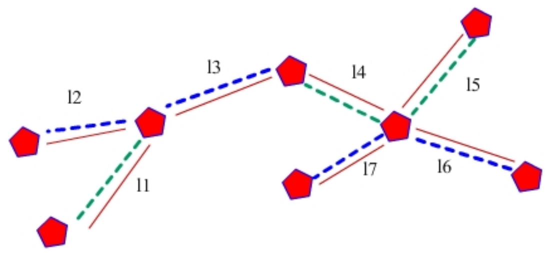

Figure 3.1: Channel Assignment using default channel

medium needs to be shared simultaneously by both the receiving nodes in order for successful reception of transmission. For explanation of our proposal we have selected the channel assignment scheme proposed in [14] shown in Fig 3.1. The reason for using this scheme is the simplicity and effectiveness of the mentioned proposal in reference to fulfillment of our requirements. The scheme mentioned allocates a single pre-defined default channel to all the nodes in the WMN. When a node comes online it first performs joining negotiations with other nodes of the network on this channel. Once the negotiations are successful other radio interfaces will be initiated on their respective channels. This channel assignment strategy only utilizes the default channel for the sake of initializing the radio interfaces and to exchange control traffic if it arrives in the future. But most importantly this strategy for channel allocations provides an inherent support for broadcast traffic. We are using default channel more effectively by transmitting Unicast Network Coded Traffic over it. In this way the packets which were to be transmited on a non default channel i.e. channel 1 are now transmitted on default channel. This saved bandwidth on channel 1, which can be used for transmitting more traffic on it. Most importantly the main advantage of network coding is that each encoded packet of a network coded transmission is a combination of two separate packets therefore by using this technique we are improving the link capacity of an intermediate link (e.g. l3 in Fig 3.1) by more than 100% (for 95-100% network coding), whereas the end nodes are to see a far better gain depending upon the topology.

3.2 Network Coding Opportunity Maximizing Algorithm

For the purpose of achieving the above mentioned improvements in WMNs we need to develop a Network Coding opportunity maximizing algorithm that is capable of availing most of network coding chances while at the same time, be efficient enough to adapt to the changes in network conditions such as traffic changes, flexibility to operate for different types of traffic and most importantly to avoid waiting needlessly for network coding to be performed at a node. The algorithms currently being used for network coding do not take into consideration the time it will take for a packet's pair to arrive at a particular intermediate node. The crucial objective for achieving network coding is to hold a packet at any given intermediate node along the path for so long that a pair packet will arrive for its encoding. In our proposed algorithm we preferred the use of inter-flow network coding to intra-flow network coding. The reason being that, the use of inter-flow network coding allows for a higher percentage of coding opportunities among transit nodes, and also this coding option results in projecting a much more practical scenario for wireless networks. There are number of issues that affect this decision as to how long a packet should be held at a node. If for instance we need to hold a packet at a given node for infinite time [4] waiting for its counterpart packet to arrive we will be achieving 100% network coding as we will be utilizing all the opportunities that are possible for pairing of packets. But in achieving this we may also delay a packet for a very long time at every node that exists in its path. This will be undesirable and will also result in issuing of recovery mechanisms at higher layers as packets will be treated as lost. On the other hand if we are to simply send a packet immediately after its has arrived and apply network coding only on packets that are available in the transit queue [2] then we will be wasting several chances that can be used for encoding and pairing of packets. We therefore need a simple and intermediate solution for this scheme for calculating the time a node should hold a packet. Our solution to this question is the use of a timer called "Holding Timer" the value of which will decide how long a packet has before it is encoded. The value of holding timer will be calculated by collecting statistics for each source-destination IP address combination for which a packet arrives at the node. By estimating the time it takes for another packet for a similar address combination to arrive we can calculate the value for which a packet should stay at a given node. Also for network coding to be performed a packet has to be routed first and its next hop MAC address should be decided before performing network coding. This is because network coding in our work will be performed on link layer and a source-destination MAC address combination will decide which packet is pair-able.

3.2.1 Algorithm

The Algorithm adopted for performing Network Coding is as follows

3.2.1.1 Upon Arrival of new packet

1. The Node receives a new packet
2. The Node compares this packet with the ones present in queue.
3. If a match is found then Node extracts the matching packet from queue
4. Node encodes both packets.
5. Node immediately sends the encoded packet for transmission.
6. If a match is not found then the Node assigns a Timer to the packet and adds it to the queue.

3.2.1.2 Upon Timer Expiry

1. If the Timer attached to a packet expires.
2. The Node removes the packet from queue.
3. The packet is sent for transmission.

3.2.2 Algorithm for Encoding

1. if Timer(i) = 0
   (a) queuedPacket := GetExpiredPacket(Queue)
   (b) TransmitPacket(queuedPacket)
   (c) end if
2. end if
3. if packetArrived = True
4. rcvdPacket := ReceivePacket()

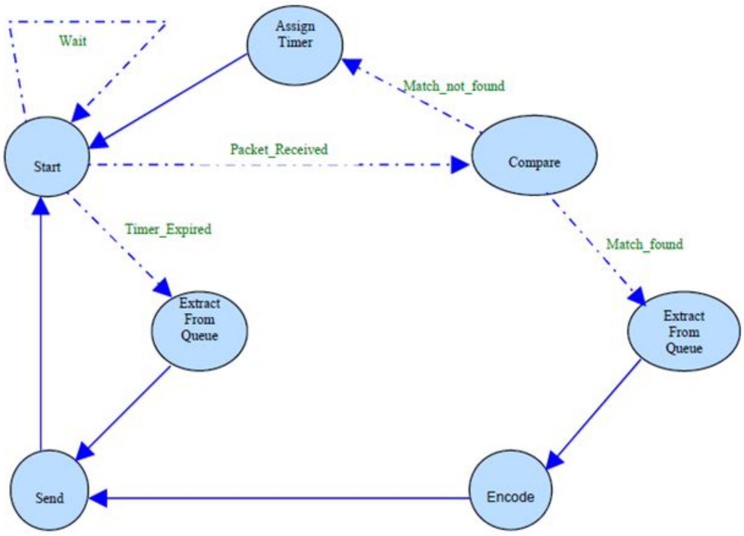

Figure 3.2: State Diagram for Network Coding Opportunity Maximizing Algorithm

(a) k = 0
(b) flag = FALSE
(c) while k <= QueueLength() OR flag = FALSE //Finds pair packets
(d) queuedPacket := Queue(k)
    i. if rcvdPacket→NextHop = queuedPacket→PreviousHop AND rcvdPacket→PreviousHop = queuedPacket→NextHop
    ii. encodedPacket := EncodePackets(rcvdPacket, queuedPacket)
    iii. TransmitPacket(encodedPacket)
    iv. flag = TRUE
    v. end if
(e) k := k + 1
(f) end while
(g) if flag = FALSE //If while loop has not paired the packet
(h) Queue→Add(rcvdPacket)
(i) StartTimer(i+1)
(j) end if

5. end if

3.2.3 Algorithm for Decoding

1. if PacketArrivedFromMAC = TRUE
2. rcvdPacket := receiveFromMAC()
   (a) if isEncoded(rcvdPacket) = TRUE
   (b) newPacket := DecodePacket(rcvdPacket)
      i. if newPacket→MAC = myMAC
      ii. AcceptPacket(newPacket)
      iii. else
      iv. DiscardPacket(newPacket)
      v. end if
   (c) else //Received Packet is not Encoded
   (d) if rcvdPacket→MAC = myMAC
   (e) AcceptPacket(rcvdPacket)
   (f) else
   (g) DiscardPacket(rcvdPacket)
   (h) end if
3. end if

3.3 Advantages

1. Regarding network coding opportunity maximizing algorithms this is the first time anyone has used the concept of timers.
2. The use of independent timers allows for inter-flow network coding.
3. Separate timers provide flexibility for traffic engineering.

3.3.1 Example

In Fig 3.3 it is shown that node A is transmitting data to node B at 2 Mbps. While node B is receiving data from node C at 4 Mbps. If we are to assign a single timer for both these overlapping flows we will have 2 packets of C in B's queue even though only 1 packet from A can be encoded. The second packet is the one that is delayed without a pair. By using a separate timer for each packet when node B receives a packet from A and C it encodes it

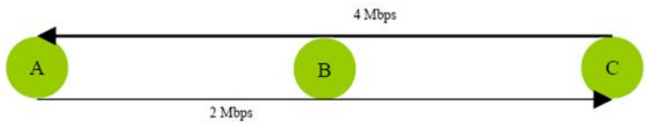

Figure 3.3: 3 Node model

and transmits it. Shortly afterwards (very shortly) node B receives another packet from C because it has a higher transmission rate. Now node B has calculated that node A will not be sending any packet soon so it assigns this 2nd packet a very small timer value instead so that it can exit the queue quickly.

3.4 Conclusion

Delaying a packet at a node gives network coding a chance to be implemented but delaying for too long will end up as performance hindrance at end nodes and may initiate packet loss recovery mechanisms. For the scope of our project however, we are not concerned with the calculation of this parameter but it will be used in our analysis for calculations of other factors such as the effect of topology (WMN) and channel assignments.

Chapter 4

Implementation

4.1 Introduction

In this chapter we discuss the practical implementations of various portions of our research work. The following discussions are related to the design decisions, such as the explanation and deployment of our proposed solutions in OPNET 14.0 Modeler as well as the advantages and disadvantages for the assignment of responsibilities to various modules. In this chapter we will discuss the following modules and their functionalities as used in the solution. Section 4.2 explains the source (Generator) module. Section 4.3 describes the Routing module. Section 4.4 explains the inner working of the Network Coding module. Section 4.5 explains the purpose and operation of the Channel Switcher module and finally section 4.6 gives a brief description of OPNET's 802.11 implementation modules

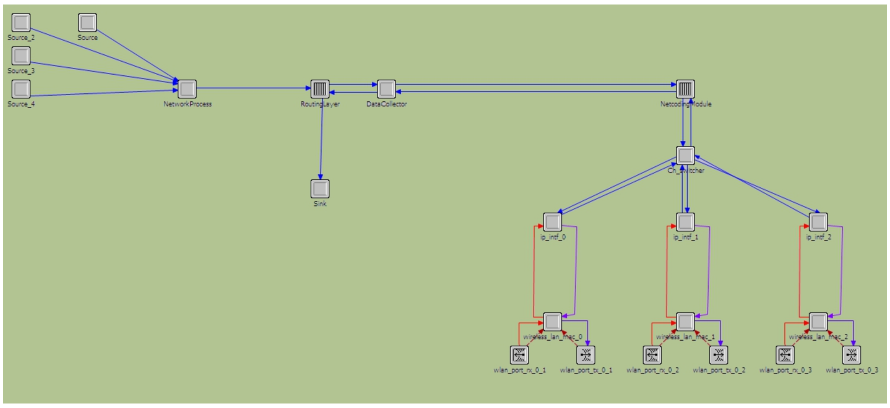

Figure 4.1: Implementation Design Structure

In Fig.4.1 the operation of the simulation starts from the source nodes which generate data for a predefined destination according to certain statistical distribution. The generated data then reaches the routing layer which reads the destination address for the packet and decides the next hop address for it. After the decision, the packet is forwarded to the netcoding layer which tries to find a pair-able packet from its queue. Since the packet is currently at the source there cannot be a pair packet for it at this node. The netcoding layer then forwards the packet to the channel switcher module which decides the radio interface is best suited for this packet and sends the packet towards that interface. At the interface the WLAN transmission protocols send the packet towards its next hop. Upon reaching the next hop the packet passes through the channel switcher module and the network coding module to reach the routing module of the neighboring node. The routing module decides the next hop, attaches the address to the packet and sends it to netcoding layer/module. Since the packet is now at a transit node it is likely that netcoding layer will find a pair packet for it. If found the netcoding layer then encodes the packet and immediately sends it to channel switcher module. The channel switcher module then sends it to the interface with the default channel to be sent as a broadcast. If the node does not find a pair at netcoding layer immediately then it is held in a queue for a specific time until which a pair packet can arrive for it. If no packet arrives for it in that time then the packet is forwarded to the channel switcher layer as a non encoded packet. The channel switcher then forwards the non encoded packets according to its specified interface. Operations for these modules are explained in further detail in the following sections.

4.2 Source Module

Source modules are used in OPNET for generating the simulated traffic. The module works by generating virtual packets which take up size only in memory buffers. This packet will be used in the underlying network scenario for simulation and analysis. The packets generated by the OPNET's generic module are fixed in size and can be generated as unformatted (raw) or formatted. The source module's purpose is to generate the packets using a predefined statistical distribution. Formatted packets can be defined through OPNET 'Packet Format Editor'. The Packet Format Editor is a simple and effective tool in defining packet formats and field types and their initial values. Our OPNET simulation's data is generated by the "simple_source" module provided in OPNET's default model library. The source module provides access to many generic data generation distributions i.e. Poisson, exponential, and logarithmic etc. These statistical distributions are flexible enough to generate traffic patterns that any kind of network may experience. For the analysis of our solution we have used Poisson traffic distribution for traffic generation because network traffic most commonly follows this pattern [16].

4.2.1 Poisson distribution

A Poisson random variable can be defined as the number of successes resulting from a Poisson experiment, where as the probability distribution of this variable is called as Poisson distribution. Poisson probability can be computed through the following formula, such that the number of successes $((\mu))$ occurs in a specific region. For a Poisson experiment the Poisson probability can be defined as

$$P\left(x;\mu\right) = \left(e^{-\mu}\right)\left(\mu^{x}\right)/x!$$

Where x is the actual number of successes that result from the experiment and e is approximately equal to 2.71828. The Poisson distribution has the following properties:

- The mean of the distribution is equal to μ.
- The variance is also equal to μ.

4.3 Routing Module

The routing module is as the name implies, used to perform IP level routing decision making and routing table maintenance. The routing module provided through OPNET's default library is capable of supporting several Ad hoc (AODV, DSR) and traditional (RIP, OSPF) routing protocols. This module is also capable of providing static routing support if needed. However this IP routing module is tightly bonded with the underlying data link module and therefore cannot be updated and therefore cannot serve our need. The conflict arises due to the unavailability of MRMC support in OPNET's default library. Once a modified data link layer containing multiple radio interfaces was attached to the existing node model the IP routing module failed to function correctly and was unable to reference multiple interfaces. We were therefore obligated to write our own module for performing the necessary task of routing. In order to perform experimentation for our solution we need the static routing option. Through static routing we selectively choose links in a network to define distinct path formations. The topology defined through these static routing paths is then subjected to different stress tests easily. Even under extreme traffic loads the paths between source and destination remains stable and contains the same links. Automated protocols such (AODV, DSR etc) are not suitable because they perform error recovery procedures when they cannot meet demands. Also, their responses are different from each other. This will introduce variance in results of different comparisons. For our static routing module we have defined the following simplified interface for maintaining routing table entries:

Destination Node	Reachable through MAC	interface#

Figure 4.2: Routing Layer Packet Format

The main function of this module is to analyze the incoming packets and decide if these packets are routable through this node or not. A filter needs to be placed at this layer because the netcoded packets are forwarded through broadcast messages. The broadcast messages have the ability to pass through the MAC layer and travel to the "Netcoded Layer/module". Once "Netcoded Layer/module" separates the two packets it does not have the ability to decide if the packet received was intended for this node or not. This short coming is due to the modular nature of the solution design (i.e. one module does not have access to other module's information). The network coding module thus sends both packets to this module. Now once the two packets are separated and received, it needs to be decided which of the packets to accept and which to discard. This decision is made by observing the destination addresses. In Fig 4.2 the "destination node" is the IP address of the node which can be reached by this node. This address will be matched with the arriving packet's destination address. If a match is found then the accompanying "Reachable through MAC" address will be attached as the next hop MAC address to the packet along with the interface number through which the Next hop can be reached. The "interface#" is the internal identification number of the radio which is on the same channel with the radio of the receiving MAC.

In the Fig 4.3 the process starts from the upper right corner of the diagram where the "Start" state is the initial state of the machine and registers and initializes counters and other variables. Upon completion the machine enters the "Read" state which reads the supplied routing file and initializes the routing table. The machine then enters the state of "Wait" where it stays until a packet arrives either from upper layer from "source module" or from lower layer "netcoding module". If packet arrives from the "source

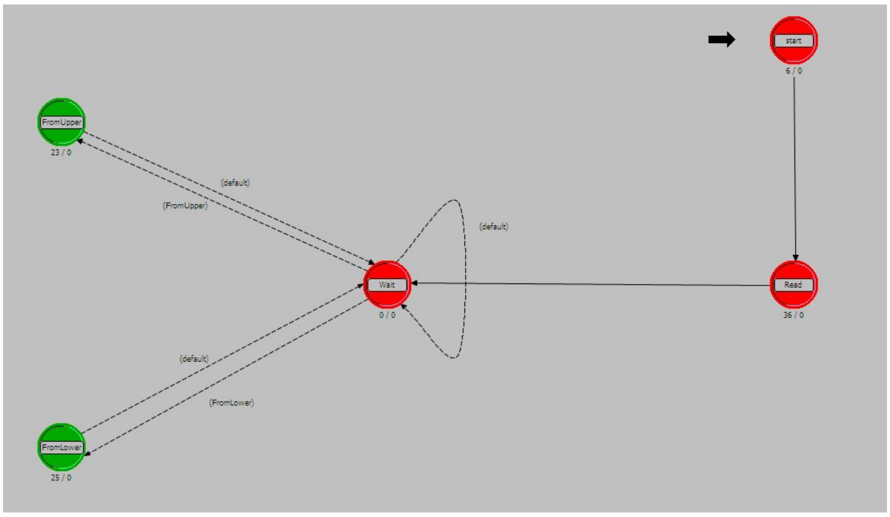

Figure 4.3: Routing Layer Custom Implementation

module" the machine triggers a transition to the state "FromUpper". This state reads the destination address from the packet and finds a next hop for it by consulting the routing table. Once the next hop is decided the MAC address is attached to the packet and the packet is sent to lower layer (netcoding module). After this the machine moves back to the "Wait" state. On the other hand if at "Wait" state a packet is received from the "Netcoding Module" the machine transitions to the state "FromLower". Here the destination address from the packet is matched with the IP address of the node itself. If matched the packet is accepted and forwarded to the sink module, otherwise the packet is routed by consulting the routing table present at the node. If the packet is found to be routable the respective MAC address is attached and the packet is forwarded to the netcoding layer.

4.4 Network Coding Module

The network coding module is the main element of our simulation model. This module contains the code for our proposed network coding opportunity maximizing algorithm. The module contains an infinite queue (not shown in Fig 4.4) which is used to hold the packets coming from the routing layer after being routed and referenced with a next hop MAC address. This module is assigned a static reference table which contains timer values for packets. This table is maintained with "Previous MAC" and "Next MAC" combinations. This Previous-Next MAC combination is crucial as it will define which packets can be netcoded at data link layer.

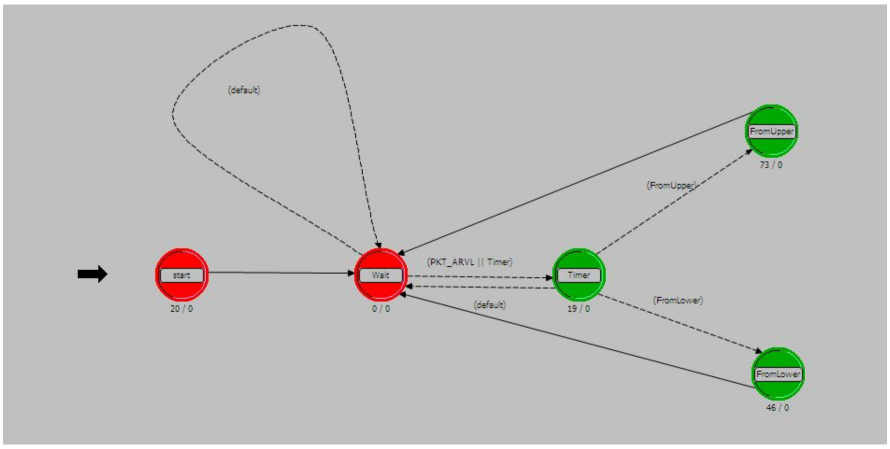

Figure 4.4: Netcoding Layer

In the Fig 4.4 the state named "Timer" is the one responsible addressing timer expiries while comparisons and encoding functions are performed in state labeled "FromUpper". The State "FromLower" simply takes the received encoded packets decodes it and sends to higher layer for processing. When a packet is received from the higher layer (Routing Module) it is first compared with all the packets present in the queue. The main comparison here is to match the "Previous Hop MAC" of the received packet with the "Next Hop MAC" of the queued packet. If matched then the "Next Hop MAC" of the received packet is compared to the "Previous Hop MAC" of the queued packet. If both conditions come true for a received packet then the matching packet is removed from queue, both packets are encoded and the encoded packet is then sent for transmission. If a match for a received packet does not exist in the queue than the packet is assigned a timer, the value of which is read from the reference table. And the packet is then inserted in the queue from where it will exit if either its pair packet arrives or if the timer assigned to this packet expires.

4.5 Channel Switcher

Channel switcher module is placed in between the netcoding layer and underlying implementations of 802.11 radios. The purpose of channel switcher module is to streamline and interface the underlying multi-radio implementations. Placement of this module relieves the higher layer modules from the tasks of coordinating and synchronizing packet transmissions.

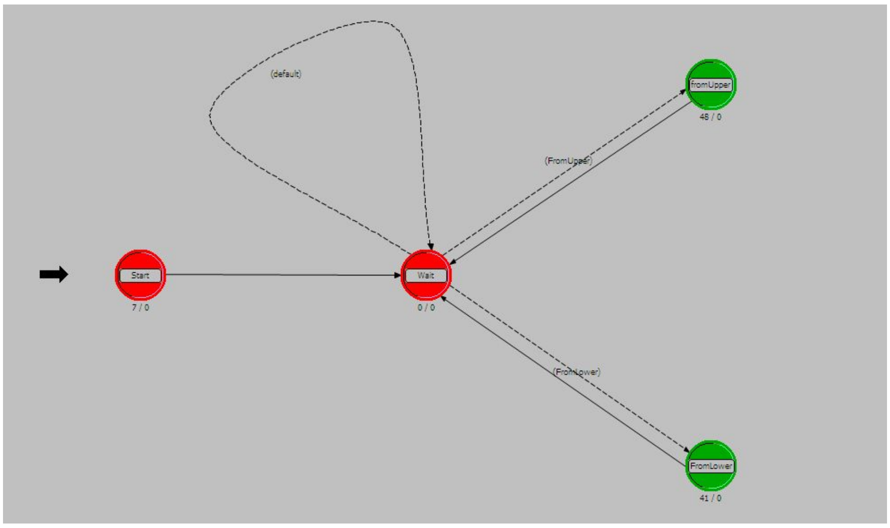

Figure 4.5: Channel Switcher

This module Fig 4.5 initializes at the "Start" state and from here the control moves to the "Wait" state. The control remains at wait state until a packet is received from the higher (Netcoding) or lower (MAC) layer. If a packet is received from higher (Netcoding) layer then the control gets transferred to the state "FromUpper". In "FromUpper" state the packet is read and from its contents the information about its "interface#" attribute is extracted. Since we are assuming that default channel is always present at these nodes, therefore the value of this attribute can be either "1", "2", "3" etc identifying the interface number, or "-1" indicating that the packet is a coded packet and must be forwarded through the interface with the default channel configured. After consulting this attribute the packet gets forwarded to the appropriate 802.11 radio for transmission and the machine moves back to "Wait" state. Similarly, if a packet arrives from underlying 802.11 radios the machine moves to state "FromLower". Here a scan is made of all the radio attached to the node. Since the MAC layer would only have forwarded this packet if it was a valid packet for this node therefore this node does not process the packet but only captures it and forwards it to the higher (Netcoding) layer.

4.6 802.11 technology's OPNET implementations

The OPNET WLAN module has the flexibility to allow it to be configured to various different flavors of the technology namely IEEE 802.11 a/b/g networks. Also any specified data rate can be fixed for transmission through the various independent radios. The module definitions are present in the default library and for the solution require some modifications. By default, the implementation of IEEE 802.11 wireless in OPNET comes with an interfacing module called as ip_arp_v4. This module is an exact implementation of the ARP layer present in most modern day devices. This module is capable of interfacing one WLAN radio at a time. Also the ip_arp_v4 module is tightly bonded and synchronized with the IP layer (ip_dispatch) process. Since the default IP module is replaced because of its short comings we also have to remove the ip_arp_v4 module. Instead of this the interfacing of the WLAN radio is done by an older interfacing module the wlan_mac_interface. The wlan_mac_interface module does not perform bi-directional synchronization with upper and lower layers. But instead only contains interfacing functions for lower layer radio control. This behavior gives us the ability to place custom designed modules on top of MAC layer and thereby forward specialized traffic.

Chapter 5

Results and Evaluation

In this chapter we are demonstrating the results obtained by the simulator for verification of our Netcoding hypothesis. Section 5.1 explores the effects of the solution for the basic three node topology. Section 5.2 discusses the effects of implementation of the solution under a random scenario. Section 5.3 performs a detailed analysis of the behavior at each link, while section 5.4 inspects the issues in channel assignments. And finely section 5.5 mentions an exclusive condition for which the behavior of performance is different. The parameters being used in the simulations for WMN are as follows:

Table 5.1: Simulation Parameters for Wireless LAN

Wireless Technology	802.11 b radios
Packet Size	512 Bytes (4096 bits)
RTS CTS	Enabled
Queue Size	Infinite
Back off Stages	9
Number Of Radios	3

5.1 Three Node Topology

The Fig 5.1 shows a basic three node network topology configuration. This configuration is used to elaborate the theory of network coding by enabling the ability of network coding in the central node.

The Fig 5.2 shows end to end traffic comparisons for the simplified three node topology. The graph shows a steady increase in throughput, in relation

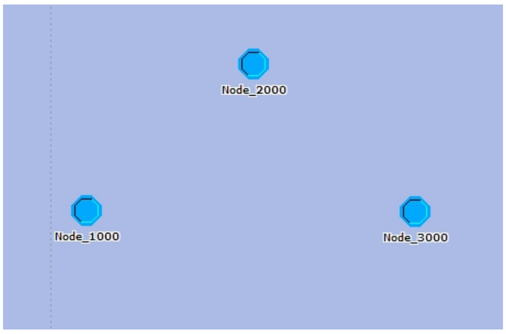

Figure 5.1: Three Node Topology

to the increase in traffic load at end nodes. Up to applied load of 1.7 Mbps, the normal routing network curve reaches its maximum point and after that there is no increase even though the traffic load at both ends is increased linearly.

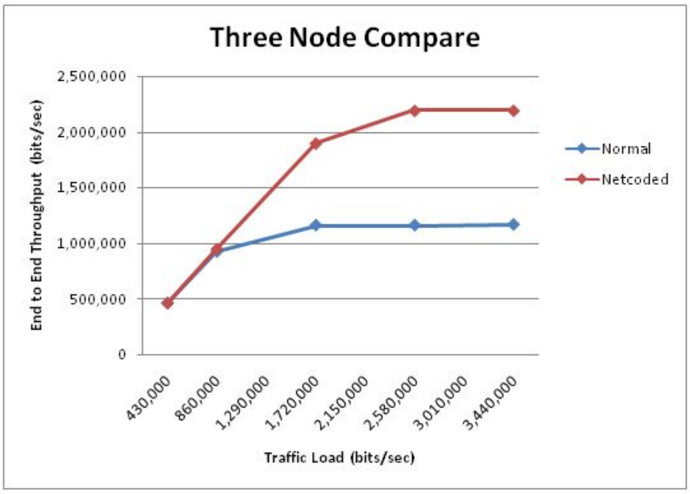

Figure 5.2: Three Node Comparison with network coding

Whereas the curve for network coding shows continues increase in the throughput even after the saturation point for its equivalent configuration. The final gain achieved by network coding is very close to the theoretical maximum gain under MR-MC configurations.

5.2 Full Mesh Test Bed

In the deployed test bed we are using the mesh mode configuration and have decided to use four paths for carrying the data as shown in Fig 5.3. These four paths are p1 (l1, l2, l3, l4), p2 (l3, l5, l6, l7), p3 (l8, l12, l13) and p4 (l9, l10, l11). We use BSF-CA channel assignment scheme [14] due to its simple deployment and the presence of a default channel at all nodes. This scheme advocates the use of the default channel for maintaining and coordinating transmissions on other channels.

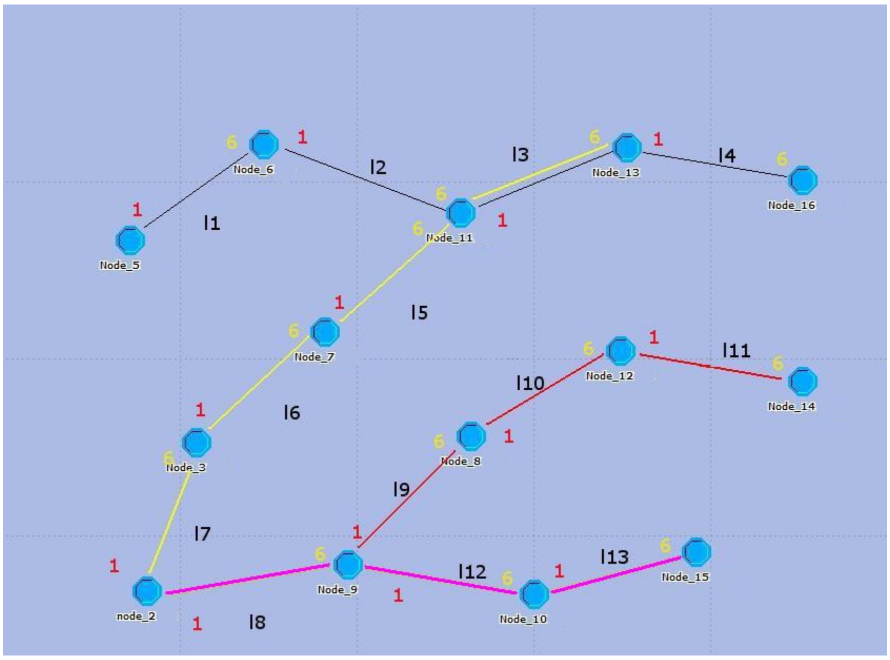

Figure 5.3: Wireless Mesh Network Topology

The graph in Fig 5.4 displays the end to end throughput comparison of the different directional flows in the test bed. The data has been collected with the load of 400 Kbps to saturate the given network.

The comparison shows that the data flows under network coding have better throughput results as compared to the data flowing under normal routing. Also there is a variation in the amount of gain that has been achieved by each of the separate flows. This variation is mainly due to the topology of the network and the channel assignment scheme used to deploy the test bed. The most prominent gain has been achieved by Flow no. 6 which is from node "2" to node "15" via path p3. Whereas the least amount of gains are achieved by Flows 5 and 7 which follow paths p1 and p4. The

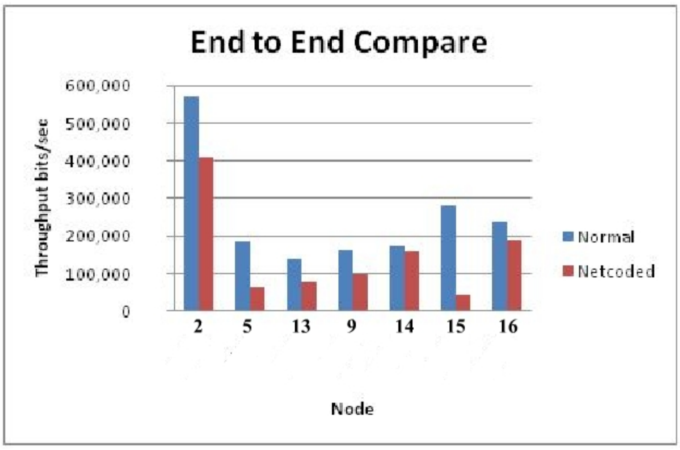

Figure 5.4: WMN End to End throughput

reason for Flow 6 to have such a large gain is because of its location at the edge of the network. As the links present in this path have fewer interfering links to contend with, more data is transmitted from node "2" to node "15" as compared to the normal routing technique where data was dropped at each of the transit links due to unavailability of transmission capacity. Here node "9" is not only a source node but also a transit node of another flow; therefore the congestion at this node became very critical. When we applied network coding a new channel became available for the transfer of transit traffic. This reduced the congestion considerably as transit data was able to move now in packets of two at a time to and from node "10" and "2". Also it can be noticed in Fig 5.3 that while all the paths are converging near the center of the network the links l12 and l13 tend to move away from the rest of the network thereby leaving congestion behind them. The throughputs at paths p1 and p4 are low because they are passing through the center of the network. The sources of these paths lie in very high congestion areas; their data never enters the network at the source, and therefore cannot be routed to the destination. In path 1 the data is getting blocked at node "6" which is the immediate neighbor of source node "5".

5.3 Evaluation of link conditions

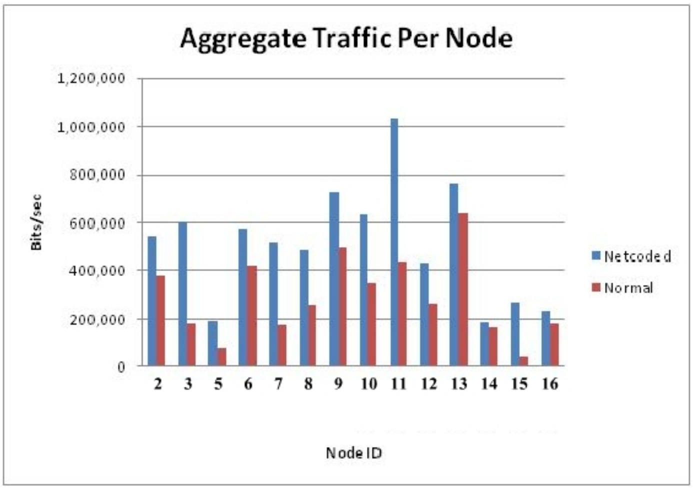

Figure 5.5: Aggregate traffic condition at all links

Regarding the traffic conditions we now consider the behavior of traffic present at all the links under saturate conditions. The Fig 5.5 shows the data received on average by each node present in the network.

The biggest gain in terms of quantity of data is seen at node 11 where the throughput has increased from 400+ Kbps to more than 1 Mbps. This is because of the presence of four separate flows traversing node 11. Each of these flows is capable of performing network coding with its pair. The second increased output is seen in node 3 where the highest gain in terms of ratio is achieved. This gain is achieved due to its special location at the edge of the network. These three nodes 3, 7, 11 constitute a bi-directional path from node 2 to node 13. Both these end nodes have achieved approx. 60% of gain which is also equivalent to the average gain experienced by the network. However theoretically the gains achieved by node 2 and 13 should be proportional to those of its transit links. This limit was not reached because of the packet drop at node 11. Although node 11 is showing considerable gain in its throughput regarding its position its gain should have been at least 1.5 Mbps. This number has been estimated for node 11 keeping in view the percentage increase in the nodes 3 and 7.

The graph in Fig 5.6 shows the comparison between the normal and net-coded traffic under varying load generations. The Fig 5.6 is generated for the aggregate of throughput at all end nodes. The figure shows a steady increase in applied load along the x-axis, while aggregate end to end demand

Figure 5.6: Aggregate End to End throughput

fulfilled is shown along y-axis. Observation of the Fig 5.6 shows in the beginning a similar traffic service rate for both normal routing and network coded transmission. However after a certain traffic load normal routing is unable to handle any more data. At this point the network coded traffic continues to increase and keeps on supporting more traffic load. After sometime the traffic under network coding also reaches a threshold and is unable to support any more traffic. For the current scenario that threshold for network coded routing is approximately 1.7 times the maximum service rate of normal routing.

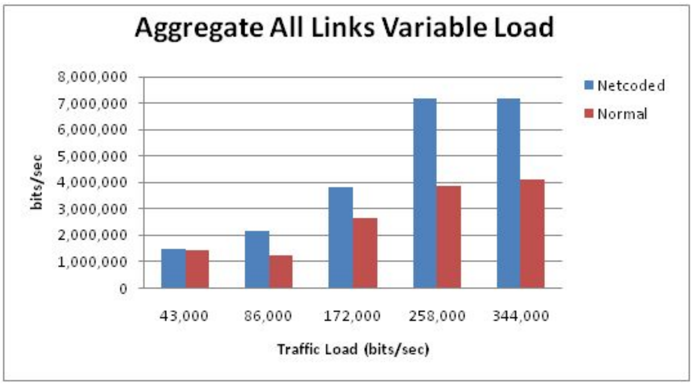

Figure 5.7: Aggregate All Links Variable Load

Comparing this with the aggregate service rate of all links we can see a steady increase in the amount of data handled by the network coding transmission. The gain here increases similar to that of Fig 5.7 until thresholds are reached for both the technologies.

5.4 Channel Assignment

Channel assignment in WMN is important dimension to consider. Efficient and interference free channel assignment can have a significant effect on the overall throughput of the network. Fig 5.8-a shows the average channel utilization as viewed by all the links present in the network. This graph indicates a high amount of activity on both channel 1 and 6 under saturation. Whereas for channel 11 the activity is sparse but contains combined data packets.

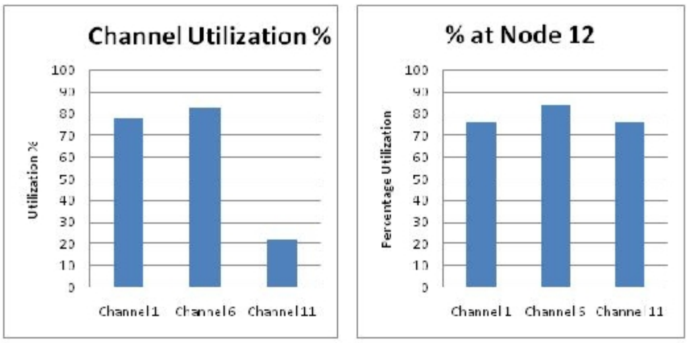

Figure 5.8: Channel Conditions (a) Avg all nodes (b) Node 12 only

Regarding the topology affect it was noticed (Fig 5.8 b) that the amount of activity witnessed by node 12 is unusually large as compared to the rest of the nodes. This suggests that node 12 is causing contention to a significant number of network nodes. This has significantly affected the throughput of many of its neighboring links. By looking at the end to end throughput results in Fig.5.6 we can see that both nodes 14 and 16 fall in the neighborhood of this node.

The comparison graph in Fig 5.9 shows the end to end throughput by switching path p4 on and off. It is observed that by turning off path p4 the throughput of nearly all the nodes of the network increase for both normal and netcoded transmissions. Also nodes 13 and 15 both see a further increased gain for netcoded transmissions. From the analysis it is clear that the presence of a single node in an important location in the network can have a severe effect on rest of the paths traveling over it or along its transmission range. In our analysis this path has been turned off only to collect the required results, and we do not endorse the exclusion of such nodes from transmissions. But since we are using 802.11 b radio technology we will always have the results as stated in the previous section Fig 5.6. However if

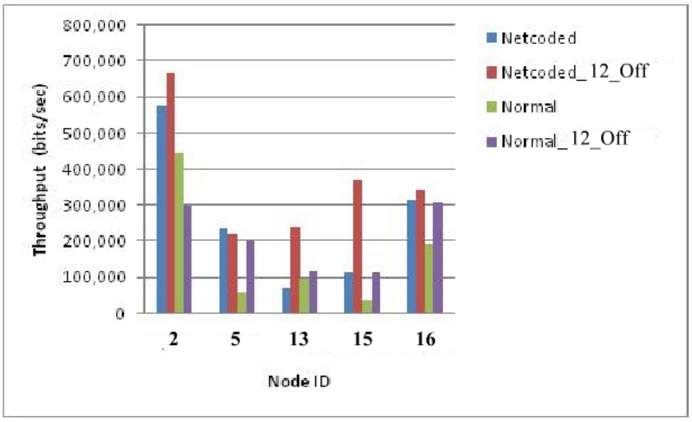

Figure 5.9: End to End Traffic Comparison by activating and deactivating path p4

we are to use 802.11 a/g radio technologies we will have multiple orthogonal channels for us to implement. For links which are found on such critical locations, we can arrange separate non interfering channels for communication. In general Network Coding, and specifically our proposed algorithm is independent of the channel assignment that exists at the physical layer and can therefore be adapted for these technologies as well.

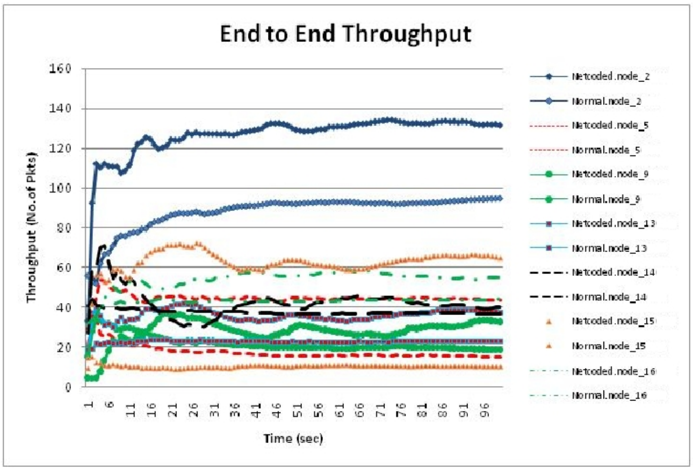

Figure 5.10: End to End throughput under saturated conditions

As a further explanation to the previous graphs this representation in Fig.5.10 shows the complete network condition against a defined time axis. This graph shows a much more detailed view of the activity present at various destination nodes.

5.5 Exclusive condition

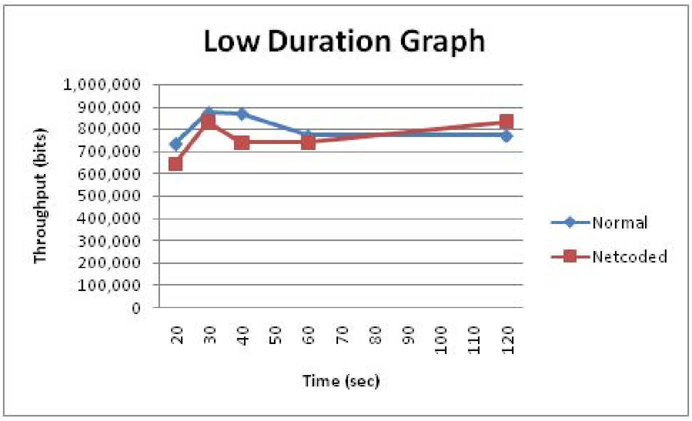

Figure 5.11: Low Duration End to End throughput graph

Throughout our discussions we have repeatedly shown that by the implementation of network coding the network throughput will increase for all networks as compared to the normal routing techniques. However for very short durations the throughput yielded by normal routing produces better results as compared to Netcoded Routing. The graph in Fig 5.11 shows the comparison between the Normal transmissions and Netcoded routing. The graph is compiled for duration of 120 seconds for saturated traffic load. This graph shows Normal routing to have slightly better results for most part of the simulation. However, during the end the throughput yield from netcoded transmissions bypasses that from normal routing. This behavior occurs because our network coded algorithm requires some time to stabilize and for packets to be "loaded" at every node in order to initiate network coding procedures. This storing of packets hampers the end to end throughput for a few seconds. The exact duration for this behavior depends upon a number of attributes namely network topology, channel assignment and type and quantity of traffic load etc.

5.6 Conclusion And Future Work

5.6.1 Conclusion

The main objective of our project is to establish that performing network coding in WMN settings will lead to a performance benefit for these networks. In order to verify this we have proposed and implemented a new Network Coding opportunity maximizing algorithm. This algorithm is capable of working on multiple channels at a time. During our analysis of the existing algorithms we found that algorithms such as COPE tend to waste several opportunities for network coding in their packet selection. Also algorithms such as NOCOCO realize this and focus on maximizing the coding performance by reducing the opportunity wastage by infinitely delaying packets.

We know that delaying a packet at a node gives network coding a chance to be implemented but delaying for too long will end up as performance hindrance at end nodes and may initiate packet loss recovery mechanisms. This led us to the concept of implementing Timers which will be attached to every packet. The value of timer will dictate for how long a packet ought to stay at a given node such that network coding can be applied effectively without any adverse affects to the system's performance.

5.6.2 Future Work

To further improve the performance of the given network a number of considerations exist. Such as

- Improve Channel Assignment strategy
- Optimize Timer duration
- Improve Network Coding Algorithm
    1. Better exploitation of Timers
    2. Intelligent selection of packets for coding

By improving channel assignment strategy for technologies like 802.11 a/g we believe that it will be possible to further improve the overall network throughput. Also our simulations mostly include the results where the Timer value has been set to maximum. By finding the optimal values for the Timers associated with each packet we may be able to reduce the end to end packet delays while keeping in view maximum network performance. Also we have shown that by turning off network coding (with Timer Value equals zero) for some flows will, in the long run, have a better impact on overall network performance. This identification of codable flows and their timer assignments is also a promising future direction for performance improvement.

Bibliography

[1] Xinyu Zhang, Baochun Li, "On the Benefits of Network Coding in Multi-Channel Wireless Networks, "in Sensor, Mesh and Ad Hoc Communications and Networks. SECON'08. 5th Annual IEEE Communications Society Conference on, 2008, pp 341–349

[2] Sachin Katti, Hariharan Rahul, Wenjun Hu, Dina Katabi, Muriel Medard, Jon Crowcroft, "XORs in The Air: Practical Wireless Network Coding," in Proceedings of the 2006 conference on Applications, technologies, architectures, and protocols for computer communications, 2006, pp 243–254

[3] Prasanna Chaporkar, Alexandre Proutiere, "Adaptive Network Coding and Scheduling for Maximizing Throughput in Wireless Networks," in Proceedings of the 13th annual ACM international conference on Mobile computing and networking, 2007, pp 135–146

[4] Bjrn Scheuermann, Wenjun Hu, Jon Crowcroft, "Near-Optimal Coordinated Coding in Wireless Multihop Networks," in Proceedings of the 2007 ACM CoNEXT conference, 2007

[5] Xiaoyan Zhu, Hao Yue and Yumin Wang, "C&M: A New Network Coding Scheme for Wireless Networks," in 2009 Fifth International Conference on Information Assurance and Security, 2009, pp 432–436

[6] Tinku Rasheed and Massimiliano Slongo, "Channel Assignment in Wireless Mesh Networks", 2008

[7] Seok-Chul Kwon, Faramarz Hendessi, Faramarz Fekri and Gordon L. Stuber, "A Novel Collaboration Scheme for Multi-Channel/Interface Network Coding," in IEEE Transactions on Wireless Communications, , pp 188-198, 2011.

[8] M. Alicherry, R. Bhatia, and L. Li, "Joint Channel Assignment and Routing for Throughput optimization in Multi-radio Wireless Mesh Net-

works," in Proceedings of the 11th annual international conference on Mobile computing and networking, 2005, 58-72

[9] H. Skalli, S. K. Das, L. Lenzini, and M. Conti, "Traffic and interference aware channel assignment for multi-radio wireless mesh networks ", in Technical Report, 2006.

[10] A. H. Mohsenian-Rad and V. W. S. Wong, Joint logical topology design, interface assignment, channel allocation, and routing for multi-channel wireless mesh networks, in IEEE Wireless Communication, p.4432-4440, December 2007

[11] B.-J. Ko, V. Misra, J. Padhye, and D. Rubenstein, "Distributed Channel Assignment in Multi-radio 802.11 Mesh Networks ", in Wireless Communications and Networking Conference (WCNC), pp. 3978–3983, 2007

[12] D. Agrawal, A. Mishra, K. Springborn, S. Banerjee and S. Ganguly , " Dynamic Interference Adaptation for Wireless Mesh Networks ", in WiMesh 06, p.33-37,2006

[13] A. Raniwala and T. Chiueh, "Architecture and algorithms for an IEEE 802.11 based multi-channel wireless mesh network ", Proc. 2005 International Conference on Computer Communication (INFOCOM), 2005

[14] K. N. Ramachandran, E. M. Belding, K. C. Almeroth, and M. M. Buddhikot, "Interference-Aware Channel Assignment in Multi-Radio Wireless Mesh Networks", Proc. IEEE INFOCOM, April 2006

[15] P. Kyasanur, J. So, C. Chereddi, and N. H. Vaidya, " Multi-Channel Mesh Networks: challenges and protocols ", in IEEE Wireless Communications, vol. 13, pp. 30–36, April 2006

[16] Victor S. Frost and Benjamin Melamed, "Traffic Modeling for Telecommunications Networks", IEEE Communications, Mar. 1994

[17] Charles E. Perkins, Samir R. Das, Elizabeth M. Belding-Royer, " Ad hoc On-Demand Distance Vector (AODV) Routing," Mobile Ad Hoc Networking Working Group,, Internet Draft,17 February 2003

[18] David B. Johnson, David A. Maltz, Yih-Chun Hu, "The Dynamic Source Routing Protocol for Mobile Ad Hoc Networks (DSR)," IETF MANET Working Group, INTERNET-DRAFT, 19 July 2004

[19] IEEE-SA Standards Board, Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications, IEEE Std 802.11b-1999, (Revised) 2003

[20] Yeung, R.W. and Zhang, Z., "Distributed source coding for satellite communications", Information Theory, IEEE Transactions on, 1111-1120, 1999

[21] http://searchnetworking.techtarget.com/definition/network-coding

[22] Chachulski, S. and Jennings, M. and Katti, S. and Katabi, D., "MORE: A network coding approach to opportunistic routing," 2006