國立交通大學

網路工程研究所

碩 士 論 文

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中文摘要

網路編碼技術利用對網路封包作加密及解密的動作來提升網路傳輸的效能。而網路編碼技術可以簡略分成兩個方向：intra-session 及 inter-session coding，在這篇論文裡，我們將重點放在 inter-session coding，由於 inter-session coding 是針對不同的一對一資料流來做編碼，所以比起 intra-session coding 的困難度相對提高。

這篇論文與之前網路編碼的理論分析研究大不相同，我們提出一個容易實作及佈建的機會式網路編碼架構並建構在 IEEE 802.16(d) 網狀網路上。不同於其他研究者提出的實作架構，我們的系統只需要路由層及媒體存取控制層的相關資訊即可發現編碼機會，而不需要額外的網路協定。由於這項優勢，我們的系統可以輕易的架構在真實世界的網路上而不需大幅的修改。

除此之外，我們指出了一項新的"延伸的隱藏終端點問題"，這個問題在使用網路編碼的網路中會很容易發生，然而，在目前常用的無線網路標準裡無法避免這個問題，所以我們提出一個頻寬保留的機制去避免此問題的發生，而這個機制是由原本 IEEE 802.16(d) 網狀模式的三方交握機制去延伸而得。我們的模擬結果展現了在應用層效能上網路編碼技術比起原本的路由機制大幅提昇，而且我們提出的頻寬保留機制可以顯著地減少延伸的隱藏終端點問題的發生及封包碰撞的次數。

關鍵字：無線網狀網路，網路編碼，IEEE 802.16，WiMAX，延伸的隱藏終端點問題

ABSTRACT

Network coding is a packet-encoding-decoding mechanism that aims to increase the data transmission efficiency of a network. The development of network coding can be roughly classified into two categories: intra-session and inter-session coding. In this thesis, we focus on the inter-session network coding because it is more challenging to improve the performances of multiple unicast flows.

Unlike other theoretical studies, we propose an easy-to-implement and easy-to-deploy network coding scheme that is based on the opportunistic approach over IEEE 802.16(d) mesh networks. As compared with previous opportunistic coding scheme, our proposed scheme need not employ an additional protocol to find coding opportunities or coding structures. Instead, only the routing and MAC-layer information are needed in our coding scheme. Using this advantage, it is easier to be realized in a real-life network as compared with the previously-proposed opportunistic coding schemes.

In addition, in this thesis we point out a new "extended hidden terminal problem," which can frequently occur in a wireless network using network coding. We will explain why the EHT problem can not be solved by currently-existing wireless network standards and propose a bandwidth reservation mechanism that is extended from the IEEE 802.16(d) mesh-mode three-way handshake procedure to prevent the EHT problem from occurring. Our simulation results show that the throughput performance of a network using our network coding scheme is better than that of a network using the traditional routing. Furthermore, our extended bandwidth reservation mechanism can reduce the EHT problem and decrease the collision occurrences largely.

Keywords: wireless mesh networks, network coding, IEEE 802.16, WiMAX, extended hidden terminal problem.

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Chapter 1

Introduction

Network coding is a packet-encoding-decoding mechanism that aims to increase the data transmission efficiency of a network. It was first proposed by *Ahlswede et al.* [2] in 2000 and has gained much attention in recent years. A typical network coding mechanism is composed of two main operations: 1) mixing distinct outgoing packets on intermediate nodes; and 2) decoding mixed packets before (or when) they arrive at their destination nodes. With carefully finding and exploiting possible network coding opportunities, network coding can reduce the number of packet transmissions required for disseminating the same amount of data, as compared with traditional routing.

The development of network coding can be classified into two categories based on the network traffic patterns that it intends to address. One deals with the traffic generated by one or more multicast flows (referred to as intra-session coding) [4], while the other deals with the traffic generated by multiple unicast flows (referred to as inter-session coding). Several previous work has proven that the intra-session coding is beneficial for multicast-flow traffic and has well studied the characteristics of this network coding class. More information for the intra-session coding can be found in *Chou et al*'s survey article in [3].

In contrast, developing an inter-session network coding scheme to effectively improve the performances of unicast flows is more challenging. In a unicast-flow scenario, each flow has only one node as its intended receiver. In such a condition, blindly coding packets is likely to waste network bandwidth to disseminate a packet to nodes that are not interested in it. As such, designing an efficient inter-session network coding scheme is difficult because such a scheme needs to consider three problems at the same time: 1) routing path selection; 2) traffic load balancing; and 3) network coding decision. [15][6][9][13] [16] formulate the inter-session network coding problem as a resource optimization problem that considers the aforementioned problems and use the linear programming technique to mathematically obtain the optimized solution to this problem.

Such theoretical studies have explored the characteristics of the inter-session coding problem; however, they do not propose feasible distributed protocols for scheduling inter-session network coding across a network. (Wang *et al.* [16] propose a theoretical distributed framework to control inter-session coding over a virtualized wireless network model. However, they do not present numerical results to evaluate the throughput and packet delay time performances of their proposed framework.) The reason is that a linear-programming-based scheduling approach usually requires high computation complexity to find the optimal scheduling, which means that a distributed algorithm developed based on this approach needs long time to converge[1] (and may not converge under highly-changing traffic pattern).

In addition, these previous studies assume that no packet collisions occur in a network; that is, they do not consider the performance degradation of an inter-session network coding scheme caused by the well-known "hidden terminal problem." The impacts of the hidden terminal problem can be discussed by two aspects. First, the packet collisions caused by the hidden terminal problem greatly reduce the overhearing capability of each node, which is fundamental to the operation of network coding. As such, the performance of a network coding scheme will be degraded if hidden terminals are present.

Second, using virtual carrier-sensing mechanisms (e.g. the RTS-CTS mecha-

[1]The convergence of a network-coding protocol is defined as: the coding and routing decisions of all nodes in a network is consistent and loop-free.

nism used in the IEEE 802.11 network) and reservation-based mechanisms (e.g. the three-way handshake procedure used in the IEEE 802.16 mesh network) to prevent the hidden terminal problem from occurring is likely to introduce time and bandwidth overheads for packet transmission. Such extra overheads also decrease the performance benefits achieved by a network coding scheme. As such, considering the impacts of the hidden terminal problem is essential to designing a good network coding scheme that aims to run in real-life networks.

To the best of the authors' knowledge, in the literature no inter-session network coding schemes that take the hidden terminal problem into account have been proposed. As such, the objective of this paper is to propose a practical inter-session network coding scheme that can be easily implemented in the real world and can improve the performance of a network. This paper has two contributions. First, in this paper we propose an easy-to-implement and easy-to-deploy inter-session network coding scheme that is based on the opportunistic approach. As compared with previous opportunistic coding schemes, such as COPE [12] and BFLY [14], our proposed scheme need not employ an additional protocol to find coding opportunities or coding structures. Instead, our proposed scheme exploits only routing and MAC-layer information to find coding opportunities and coding structures. Using this advantage, it is easier to be realized in a real-life network, as compared with the previously-proposed opportunistic coding schemes.

Second, in this paper we point out a new "extended hidden terminal problem," (denoted as the EHT problem) which can frequently occur in a wireless network using network coding. (In this paper, we call such a network a "network-coded wireless network.") The EHT problem can result in the failure of network-coded packet decoding, thus significantly decreasing the performance of a network-coded wireless network. In Chapter 5, we will explain why the EHT problem cannot be solved by the currently-existing wireless network standards, such as the IEEE 802.11(b) and the IEEE 802.16(d) specifications. In addition, we also propose a bandwidth reservation mechanism that is extended from the IEEE 802.16(d) mesh-mode three-way handshake procedure to prevent the EHT problem from occurring.

Although our mechanism proposed in this paper is specific to the IEEE 802.16 mesh network, it can be used as an example protocol design to eliminate the EHT problem for other network-coded wireless networks.

The rest of this thesis is organized as follows. In Chapter 2, we present related work and explain the different of our work and other previous work that opportunistically performs network coding. In Chapter 3, the background of the network coding theory and the IEEE 802.16(d) mesh network are introduced. In Chapter 4, we propose our opportunistic inter-session network coding scheme and explain the detail issues of practical implementation. In addition, we explain the EHT problem and provide our solutions in Chapter 5. Furthermore, we present the simulation results and performance evaluations in Chapter 6. Finally, we propose possible extensions to our work in Chapter 7 and conclude the thesis in Chapter 8.

Chapter 2

Related Work

In the literature, several papers ([15][6][9][13][16]) have theoretically studied the inter-session network coding problem. In these studies, the authors modelled the inter-session network coding problem as a resource allocation problem that considers a number of factors, such as coding chances, queue size of nodes, and link capacity, and then find the optimal allocation for such a problem using linear programming-based approaches. Although these previous studies have discussed the characteristics of the inter-session network coding problem, turning their mathematical results into feasible distributed real-life protocols is difficult due to the huge computation complexity required by the linear-programming-based mathematical operations.

On the other hand, another track of inter-session network coding research is *opportunistic coding*, which aims to develop sub-optimal yet easy-to-implement and easy-to-deploy network coding schemes for unicast-flow wireless networks. In [12], Katti *et al.* implemented the COPE coding scheme over a real-life IEEE 802.11(b) wireless network. COPE performs network coding operation in an opportunistic manner, and its main operations are described below.

Using COPE, each node periodically broadcasts a *"reception report"* message to its neighboring nodes. A reception report message of node i contains the information of packets that node i currently possesses. Based on received reception report messages, a network node maintains a packet information pool that records which packets are currently possessed by its neighboring nodes. Each time when a node is going to transmit a data packet, it first looks up the packet information pool to check whether a coding opportunity exists or not. If so, this node mixes several packets that can be coded together to form a new network-coded packet and sends this network-coded packet out. If not, it defers the transmission of this packet until the maximum allowed waiting time has elapsed. In this condition, the packet should be transmitted immediately.

COPE uses the simple protocol explained above to help a node find coding opportunities within its one-hop neighborhood, i.e., using COPE a node is only capable of finding coding opportunities in chain-based or start-based topologies. In [14], Omiwade *et al.* propose the BFLY scheme that allows nodes to find coding opportunities in a butterfly topology. The operation of BFLY is briefly described here.

Using BFLY, each node i should periodically broadcast a hello message to its neighboring nodes. A hello message of node i contains node i's neighboring node list and a neighbor vector describing the relative neighboring relationship (regarding the butterfly structure) of node i and its neighboring nodes. As such, each node can maintain all of its possible butterfly coding structures based on the hello messages advertised by its neighboring nodes. Upon transmitting data packets, a node can check whether its outgoing packets can generate coding opportunities over the maintained butterfly structures. If so, the node mixes these packets to form a network-coded packets and then transmits the packet out.

The idea of BFLY is similar to our proposed network coding scheme, which can find coding opportunities in chain-based, star-based, and butterfly-based topologies. However, as compared with COPE and BFLY, our proposed network coding scheme has several advantages. First, COPE and BFLY need to employ additional protocols to exchange the information of either packets possessed by neighboring nodes (in COPE) or butterfly-structure relationship of neighboring nodes (in BFLY). In contrast, our proposed network coding scheme utilizes only routing and MAC-layer neighborhood information to find coding opportunities and thus need not employ any additional protocols to operate. As such, the bandwidth and time overheads generated by such additional protocols can be eliminated. In addition, because the operation of our proposed coding scheme need not rely on additional protocols, the implementation complexity and deployment cost of our proposed scheme is greatly lower than that of COPE and BFLY.

Second, to the best of the authors' knowledge, the coding schemes proposed in the previous studies, such as COPE and BFLY, do not consider the performance degradation caused by the hidden terminal problem. As such, these schemes can only work well over an interference-free wireless network, which is uncommon in the real world. In contrast, our work takes the hidden terminal problem into account and studies its impact on the performances of inter-session network coding.

Besides, we point out a new extended hidden terminal problem that only takes place in a wireless network-coded network. With the presence of hidden terminals and extended hidden terminals, the goodputs obtained by application programs in a wireless network-coded network can be greatly decreased (even much lower than those in a traditional routing-based network). To address this problem, our proposed network coding scheme employs an *"extended three-way handshake procedure (ETHP)"* to eliminate the HT and EHT problems in a wireless network-coded network. Our simulation results show that our proposed network coding scheme can effectively eliminate the occurrence of HTs and EHTs and significantly increase the application goodputs of of a wireless network-coded network.

Chapter 3

Background

This chapter gives the background of this thesis which includes two main aspects: Network Coding and IEEE 802.16(d) networks. In the following, we shall briefly introduce the fundamental knowledge about each territory.

3.1 Overview of Network Coding

Network coding is a new paradigm to increase the utilization of both wired and wireless networks. The core idea of network coding is to allow information mixing in the intermediate nodes. A receiver sees these mixed-up packets and deduces from the messages which were originally intended for the data sink. The fundamental concept of network coding was first stated by [2], which demonstrated the famous butterfly example. Consider the topology in Fig. 3.1, where source node S1 wants to send packet P1 to destination T1, and source node S2 wants to deliver packet P2 to destination T2. Using the traditional techniques of packet forwarding, packet P1 and P2 should be passed through R1 and R2, and finally be delivered to the destination T1 and T2. As a convention, we use the term "routing" to describe this traditional solution. Unfortunately, there will be a bottleneck at link between R1 and R2 in the routing solution if each link capacity is only one packet.

In [2], the network coding scheme sloves this problem by mixing packets P1 and P2 at node R1. A simple network coding mixing method is using the "exclusive

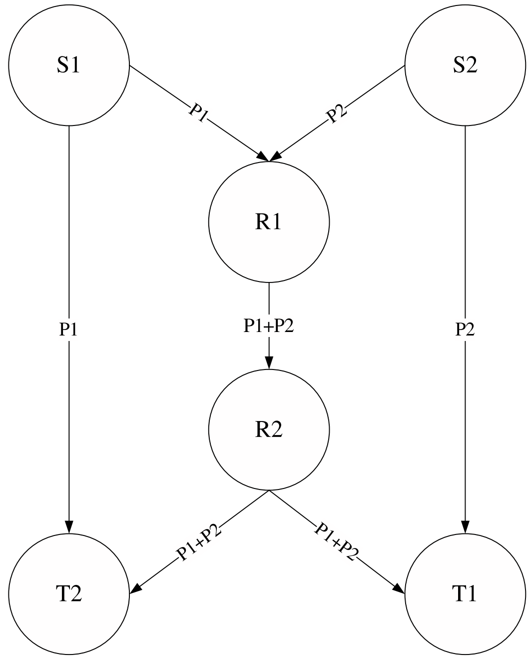
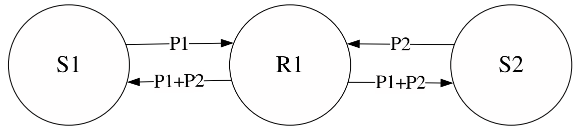


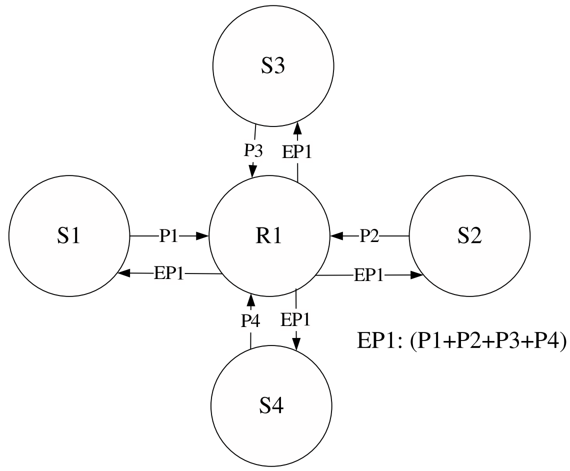
Figure 3.1: The general butterfly example

or" (XOR) to encode two packets. In a butterfly topology like Fig. 3.1, at first time node S1 and S2 broadcast P1 and P2. Node R1 gets both two packets and node T1 and T2 also receive P2 and P1 respectively. Each broadcasting casts one transmission time. Also, node R1 and R2 broadcast the encoding packet "P1+P2", and node T1 and T2 can decode to the packets they want. Therefore, we can reduce the transmissions from six to four and produce a coding gain of $\frac{6}{4} = 1.5$.

As mentioned in Chapter 1, most of the network coding researches after [2] focus on the theoretical studies rather than the pratical approaches. Since the theoretical studies have explored the characteristics of network coding, they do not propose a feasible protocol and some assumptions of them can not be ignored in real-world networks. On the contrary, the pratical implementations like [12][14] are based on the opportunistic approach. They need to find coding opportunities or coding structures to proceed the network coding scheme. The coding structures includes chain, star and butterfly structures. The butterfly structure is shown as in Fig. 3.1 and its coding benefits are explained above. In addition, the chain and star (or cross) structure are depicted in Fig. 3.2. At the chain structure example as Fig. 3.2 (a), node S1 and S2 want to deliver P1 and P2 to each other through node R1. The network coding opportunitie is obvious where node R1 encodes (P1,P2) and



(a) The General Chain Example



(b) The General Star Example

Figure 3.2: The chain and star coding structures

broadcasts to each sinks. The Structure in Fig. 3.2 (a) can reduce the transmissions from four to three and generate a coding gain of $\frac{4}{3} = 1.33$ by using network coding.

Similarly, the star structure shown as Fig. 3.2 (b) is a kind of variations of the chain structure. On the other hand, many chain structures comprised a star structure. Fig. 3.2 (b) is also named the cross structure which forms by two chain structures. In short, if there is only one forwarding node in the center (like R1) and N source nodes transmit data to each opposing side across the forwarding node, the central node can encode N packets and broadcast once. The coding gain of the star structures can be computed as $\frac{2N}{N+1}$ if all destinations can decode successfully.

3.2 Overview of IEEE 802.16(d) Network

IEEE 802.16(d) standard (also known as WiMAX) [1] is a promising next-generation technology for the future FBWA[1] systems. In this standard, the MAC layer is structured to support multiple PHY-layer specifications, each suited to a particular operational environment. In addition, the standard also supports IP-based network

[1]Fixed Broadband Wireless Access.

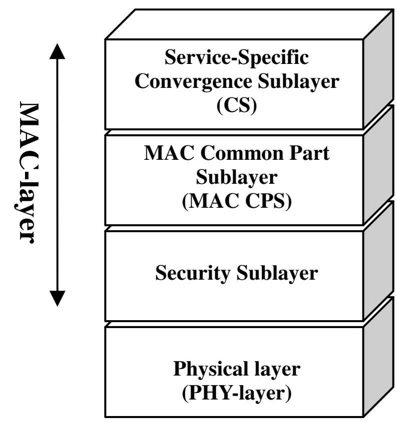


Figure 3.3: The IEEE 802.16(d) protocol layering

architecture (i.e. all WiMAX nodes are layer-3 devices).

Two operating modes are defined in the standard: the PMP mode and the Mesh mode. Firstly, the PMP mode is designed to resolve the last-mile problem in traditional wired environment. When this mode operates in the 10-66 GHz licensed bands, its line-of-sight (LOS) applications offer data rates grater than 120 Mbit/sec. Secondly, the Mesh mode is an extension mode to support the infrastructure of Wireless-MANs. In addition, this mode operates in the licensed bands below 11 GHz and uses orthogonal frequency division multiplexing (OFDM) technology. Therefore, it can support non-LOS (NLOS) applications. Compared with the traditional 802.11-based mesh network, the WiMAX Mesh Network is able to provide higher throughput and larger coverage. *In this thesis, we will only focus on the Mesh operating mode.*

Fig. 3.3 illustrates the reference mode defined in the standard. The MAC layer consists of three sublayers, which are the Service-Specific Convergence Sublayer (CS), the MAC Common Part Sublayer (MAC CPS) and the Security Sublayer. The PHY layer adopts single-carrier technology to support LOS applications, and uses OFDM technology to support NLOS applications.

The functionality of each MAC sublayer is presented in the following. In the Service-Specific Convergence Sublayer, the major function is classifying the data from upper layer, and then associating them to appropriate MAC connections. In the

MAC Common Part Sublayer, all of the main Mesh operations such as network entry process and network synchronization are specified. The Security Sublayer provides the subscribers with privacy across FBWA system by encrypting the connections between BS and SS. *In this section, we regard the MAC CPS as MAC layer directly and only focus on this sublayer.*

3.2.1 Definitions of Teminologies

In the following, some terminologies used extensively in the standard are introduced.

- Mesh BS and Mesh SS

  The station that has a direct link to backhaul services (i.e. the Internet) outside the WiMAX Mesh network is termed as a Mesh BS[2]. Other stations are termed as Mesh SSs[3].

- Sponsoring Node, Candidate Node and Registration Node

  These three types of nodes are only used in network entry process. The Sponsoring Node assists a new node in entering the WiMAX Mesh network during the network entry process. The new node involved in this process is termed as a Candidate Node. The Mesh node which assigns Node IDs to the new nodes is termed as a Registration Node.

- Link

  A link is a directional mapping between two MAC peers for either control messages exchange or traffic flows transmission. It is identified by an 8-bit link identifier (Link ID) and used to construct the connection identifier (CID). Note that a link is a logical communication between two Mesh nodes in their MAC-layer views.

- Connection

[2]Base Station
[3]Subscriber Station

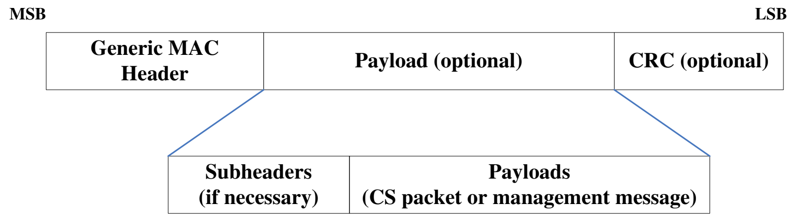


Figure 3.4: The mac PDU format

A connection, instead of a link, is a unidirectional mapping between two MAC peers. For a specified link, many different connections are constructed by either the various QoS parameters or the different network logical ID (i.e. A link can construct multiple connections). In addition, a connection is identified by a 16-bit connection identifier (CID). Note that a connection is a logical link between two Mesh nodes in their MAC-layer views as a link.

- Neighbor, Neighborhood and Extended Neighborhood

  The station that a node has a direct link connecting with is defined as the node's neighbor. Neighbors of a node form a neighborhood, and they are considered to be one-hop away from this node. An extended neighborhood covers not only a node's neighbors but also the neighbors of those being in the node's neighborhood.

MAC PDU

As shown in Fig. 3.4, the MAC PDU comprises a 6-byte generic MAC header, a variable-length Payload and an optional 4-byte cyclic redundancy check (CRC). In addition, there are zero or more subheaders and the payloads carried in the Payload field. The payloads, instead of the Payload, are used to carry either the data from upper layer (such as IP packet) or the MAC management messages.

The major functions of generic MAC header, subheader and CRC are described as follows. The generic MAC header indicates what the subheaders are carried in the Payload field and whether the CRC is appended. The extra information, such as the payload fragmentation state or the transmitting Node ID, are specified

Table 3.1: The mac management messages used in the mesh mode

Message Name	**Message Description**
REG-REQ	Registration Request
REG-RSP	Registration Response
SBC-REQ	SS Basic Capability Request
SBC-RSP	SS Basic Capability Response
MSH-NCFG	Mesh Network Configuration
MSH-NENT	Mesh Network Entry
MSH-DSCH	Mesh Distributed Schedule
MSH-CSCH	Mesh Centralized Schedule
MSH-CSCF	Mesh Centralized Schedule Configuration

in the corresponding subheaders. The CRC covers the generic MAC header and the Payload to achieve error-protecting. Table 3.1 lists the important management messages used in the Mesh mode.

Frame Structure

The Mesh frame structure is illustrated in Fig. 3.5 and Fig. 3.6. Only time division duplex (TDD) is supported in the Mesh mode. A Mesh frame consists of a control subframe and a data subframe. In addition, the control subframe is divided into transmission opportunities (TxOpps); the data subframe is divided into minislots. There are two kinds of control subframes which named the network control subframe and the schedule control subframe. The former is used for network configuration, new node entry and network synchronization. The latter is used to exchange the coordinated scheduling information. In the following, we shall introduce the control subframe and data subframe in detail.

The length of a control subframe is a fixed value of MSH-CTRL-LEN TxOpps. Each TxOpp comprises 7 OFDM symbols. A control subframe is either a network control subframe or a schedule control subframe. In the network control subframe,

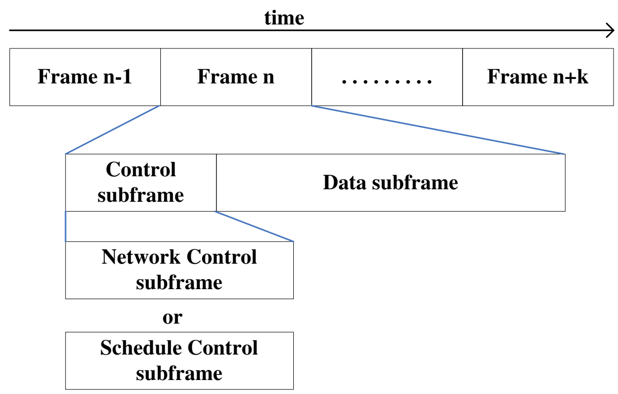


Figure 3.5: The mesh frame structure

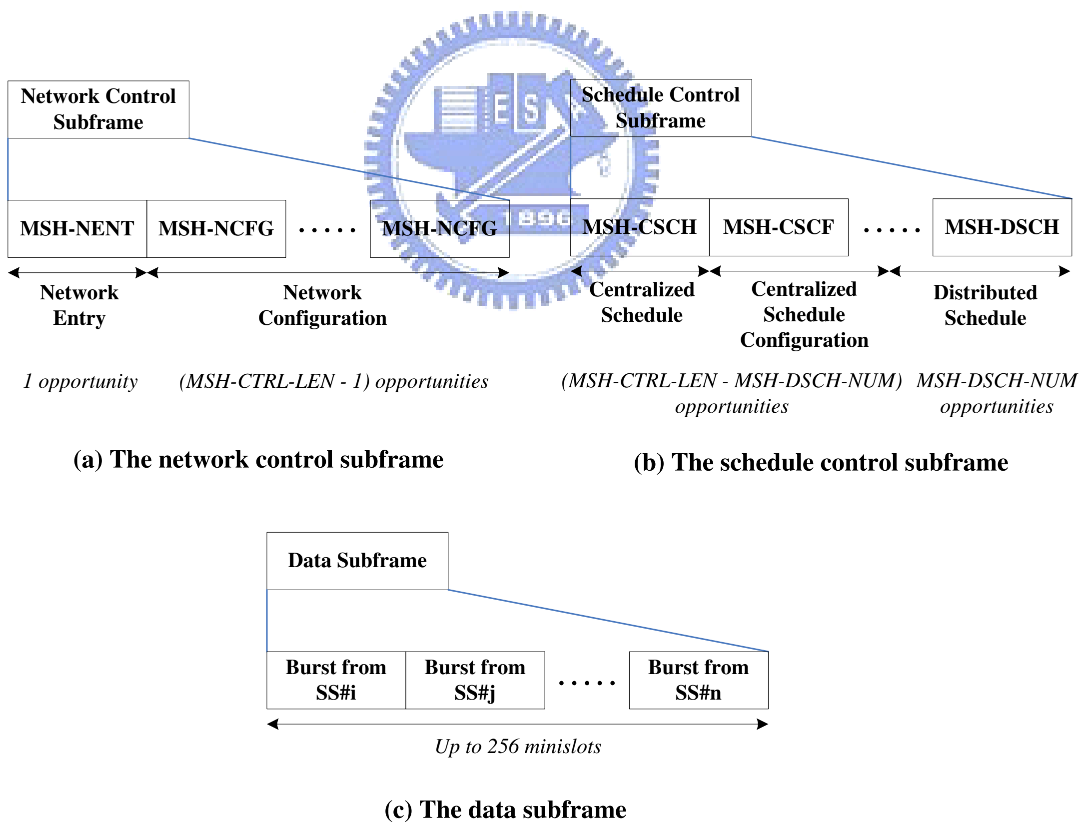


Figure 3.6: Mesh subframes in detail

one TxOpp is reserved for the network entry process, and the (MSH-CTRL-LEN - 1) TxOpps following the reserved one is used for the network configuration. On the other hand, in the schedule control subframe, MSH-DSCH-NUM defines the number of MSH-DSCH TxOpps per schedule control subframe. In addition, during this subframe, the first (MSH-CTRL-LEN - MSH-DSCH-NUM) TxOpps are reserved for centralized scheduling. The remainder is allocated for distributed scheduling. The parameters about the Mesh frame format are carried in the Network Descriptor of NCFG message.

The data subframe is divided into (up to) 256 minislots. In this subframe, the coordinated scheduling data and the uncoordinated scheduling packets will take place. A scheduled allocation consists of one or more minislots.

3.2.2 Network Entry Process

A new Mesh SS node shall perform the network entry process to join the Mesh network before starting a scheduled transmission. In the following, we elaborate this procedure step by step.

1. First of all, the new node listens to the ongoing transmissions in the air and searches for MSH-NCFG messages to synchronize coarsely with the Mesh network. In the meantime, the new node shall build a physical neighbor list according to the information carried in the MSH-NCFG message.

2. When enough information is acquired, the new node selects a Sponsor Node from its neighbor list, and it becomes a Candidate Node. Moreover, the new node shall synchronize its time to this Sponsor Node by the assumption of no propagation delays.

3. By exchanging MSH-NENT and MSH-NCFG messages, the Candidate node and the Sponsor node establish a temporary schedule named Sponsor Channel.

4. Through the Sponsor Channel, the Sponsor Node assists the Candidate node to finish the basic capabilities negotiation and authorization with the BS.

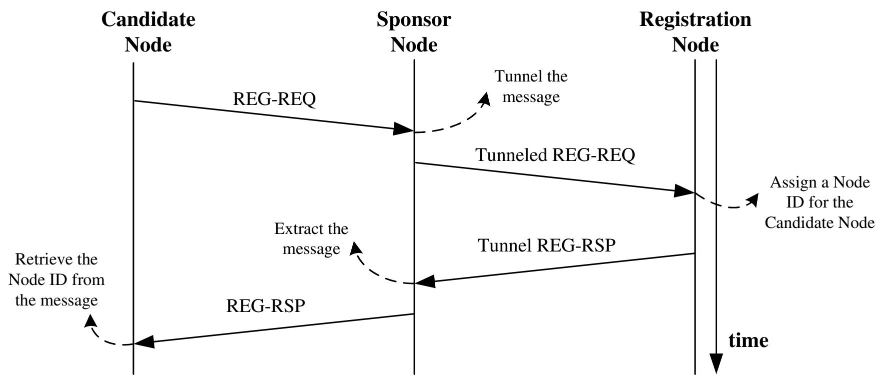


Figure 3.7: The registration process

Afterward the Candidate node shall perform the Registration Process to obtain its Node ID, as shown in Fig. 3.7.

5. At the beginning of the Registration Process, the Candidate Node transmits a REG-REQ message to register with the Registration Node via its Sponsor Node. Upon receiving the REG-REQ message the Sponsor Node tunnels the message by prepending a tunnel subheader, a UDP header and an IP header. The tunneled message is sent to the Registration Node, which can optionally be co-located with the Mesh BS. Upon the reception of this message, the Registration Node assigns the Node ID of the Candidate Node and replies the tunneled REQ-RSP message. When receiving this tunneled message, the Sponsor Node shall extract the message and forward the REQ-RSP message to the Candidate Node. Eventually, the Candidate Node obtains its Node ID and accomplishes the Registration Process.

6. While finishing the Registration Process, the Candidate Node continues to acquire an IP address using DHCP and retrieves the current system time via the protocol defined in IETF RFC 868. Furthermore, it downloads the file containing operational parameters by using TFTP.

7. Finally, after completing above processes, the Candidate Node closes the Sponsor Channel and becomes a functional node.

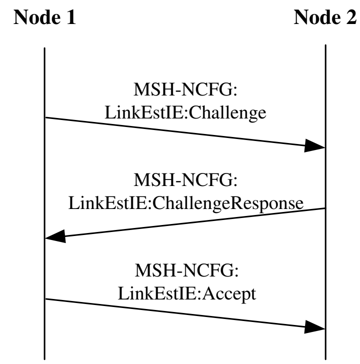


Figure 3.8: The procedure of the link establishment process

3.2.3 Link Establishment Process

The Link Establishment Process is a three-way handshake procedure that performs a simple authentication and establishes two unidirectional links between a pair of neighboring functional nodes. Before the new node can transmit data packets, it has to establish links with its neighbors by a Link Establishment Process. In the following, we take an example to explain this procedure step by step and illustrate it in Fig. 3.8.

1. At first, the Node 1 begins the Link Establishment Process by sending a challenge message to Node 2.

2. While receiving the challenge message, the Node 2 will try to authenticate Node 1. If the authentication is successful, the Node 2 will create a link from itself to Node 1 and then send back a response challenge message carrying this link value to Node 1.

3. Similarly, when receiving the response challenge message of Node 2, the Node 1 first performs the authentication for Node 2. If the authentication is successful, a link is established from itself to Node 2. Finally, the Node 1 will transmit an acceptance message carrying the link value to Node 2. This message also indicates that the Link Establish Process has been finished. Therefore, two one-way links are created safely between the Node 1 and Node 2.

3.2.4 Network Synchronization

In the WiMAX Mesh network, functional nodes periodically broadcast MSH-NCFG messages to exchange network configuration information with their neighbors. This procedure is named the Network Synchronization. Moreover, the transmission timing of the synchronized messages is determined by a pseudo-random algorithm. The algorithm works without explicit negotiation and is completely distributed, fair and robust. In the following, we introduce the Network Synchronization and the pseudo-random algorithm respectively.

Network Synchronization

In the Network Synchronization, a node does not broadcast its exact next **Xmt Time** (i.e., the time slot when a node transmits its MSH-NCFG messages). In contrast, a node advertises an interval covering the actual **Xmt Time**. The interval is only composed of a 5-bit **Next Xmt Mx** and a 3-bit **Xmt Holdoff Exponent** for reducing the signaling overhead. Consequently, as shown in Fig. 3.9, the neighbors only know the interval that covers the actual next transmission time (**Next Xmt Time**) of the sender. The interval of the **Next Xmt Time** can be computed as follows:

$$2^{XmtHoldoffExponent} \cdot NextXmtMx < NextXmtTime \le 2^{XmtHoldoffExponent} \cdot (NextXmtMx + 1)$$

A node broadcasts not only its own **Next Xmt Mx** and **Xmt Holdoff Exponent** but also these two values of all its one-hop neighbors. Therefore, every regular node possesses the scheduling information within its extended neighborhood.

Pseudo-random algorithm

When a node is able to transmit the MSH-NCFG message, it shall schedule its next transmit time of MSH-NCFG message. The determination of the **Next Xmt Time** is accomplished by a pseudo-random algorithm according to the information deriving from its neighbors. In the following, we define the terminologies used in

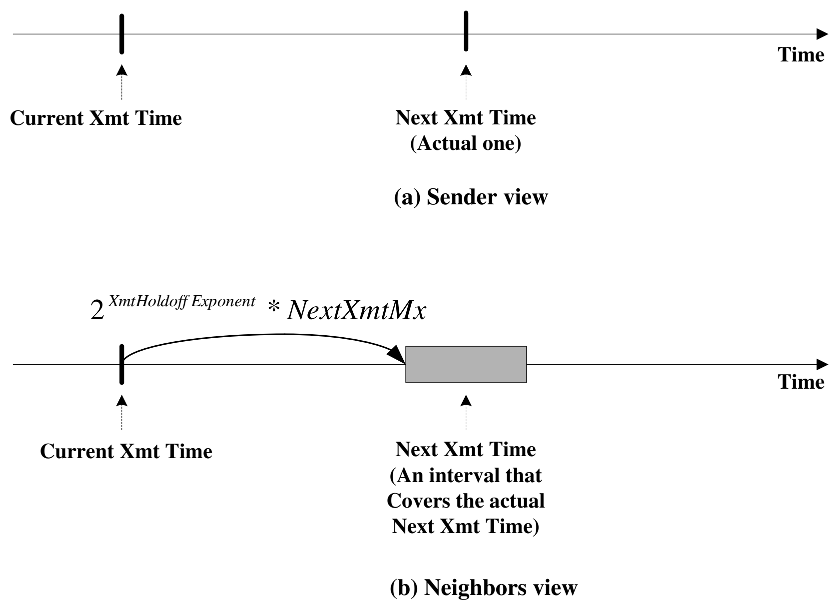


Figure 3.9: The Next Xmt Time in both sender's and neighbors' views

this algorithm.

- **Xmt Holdoff Time**

  This is the number of MSH-NCFG TxOpps after **Next Xmt Time**. In this period, a node is not eligible to transmit any MSH-NCFG packet. This value can be computed as follows:

$$XmtHoldOffTime = 2^{XmtHoldoffExponent+4}$$

- **Earliest Subsequent Xmt Time**

  The Earliest Subsequent Xmt Time indicates the earliest time when a node is capable of competing the candidate TxOpp, and it can be obtained by

$$EarliestSubsequentXmtTime = 2^{XmtHoldoffExponent} \cdot NextXmtMx + XmtHoldoffTime + 1$$

- **Temp Xmt Time**

  Temp Xmt Time starts from the TxOpp which is equal to current **Xmt Time** plus **Xmt Holdoff Time**. This time is when this node can compete for TxOpps. If this node does not win the election, Temp Xmt Time will be set to next TxOpp until this node wins.

By definition, the node only competes for TxOpps later than the current **Xmt Time** plus the node's **Xmt Holdoff Time**. That's when the **Temp Xmt Time** starts from. Before the election begins, Mesh nodes shall determine what neighbors are eligible to compete for the **Temp Xmt Time**. There are three conditions in the following. If a neighbor meets one of them, the neighbor node is eligible.

1. The **Next Xmt Time** interval of the neighbor includes the **Temp Xmt Time**.
2. The **Earliest Subsequent Xmt Time** of the neighbor is less than or equal to the **Temp Xmt Time**.
3. The schedule of the neighbor is unknown.

When a Mesh election for the **Temp Xmt Time** is held among the sender and its eligible neighbors, a pseudo-random mixing number $f_{(Node\ ID,\ Temp\ Xmt\ Time)}$ is computed for each of the nodes involved in the election. If $f_{(Sender's\ Node\ ID,\ Temp\ Xmt\ Time)}$ is the greatest among all numbers, the sender sets its **Next Xmt Time** equal to **Temp Xmt Time**, i.e. it can use the winning TxOpp to transmit its MSH-NCFG message. Otherwise, the **Temp Xmt Time** is advanced and the algorithm is repeated until the node wins a TxOpp. Note that the fairness is ensured by the algorithm and the seeds (**Temp Xmt Time**) are different for every TxOpp.

After the **Next Xmt Time** is decided, it is converted into the corresponding **Next Xmt Mx** based on the specified **Xmt Holdoff Time**. Then the **Next Xmt Mx** and the specified **Xmt Holdoff Time** are added to the outgoing MSH-NCFG message.

3.2.5 Distributed Scheduling

In the WiMAX Mesh network, the network resources (i.e. minislots) may be allocated by three different scheduling modes: the centralized scheduling mode, the distributed scheduling mode and combination of above two modes. In this thesis, we only focus on the distributed scheduling.

In the distributed scheduling mode, schedules are established in a distributed way in which all the Mesh nodes involved are regards as peers (including the Mesh BS). The schedules are set up by a three-way handshake mechanism, which ensures the established schedules to be collision-free within the extended neighborhood. In the following we describe the distributed scheduling mode in more detail.

The distributed scheduling mode is divided into two operational modes, namely the coordinated mode and the uncoordinated mode. In both coordinated and uncoordinated modes, schedules are established between two nodes using a three-way handshake mechanism by the MSH-DSCH message exchange. In the coordinated distributed scheduling mode, the MSH-DSCH messages are transmitted over TxOpps in the schedule control subframe without collisions. In contrast, in the uncoordinated distributed scheduling mode, the MSH-DSCH messages can only be exchanged in the data subframe. Therefore, collisions may occur in the uncoordinated distributed scheduling mode. *In this thesis, we only discuss the coordinated distributed scheduling mode.*

There are four kinds of information elements (IEs) that can be included in a MSH-DSCH message. The MSH-DSCH:SchedulingIE carries the coordinated distributed scheduling information: **Next Xmt Mx** and **Xmt Holdoff Exponent**. The MSH-DSCH:RequestIE is used to convey resource requests on a specified link with the demand expressed in minislots. The MSH-DSCH:AvailabilityIE indicates free minislots ranges of the requesting node, which one MSH-DSCH:RequestIE can be corresponded to multiple MSH-DSCH:AvailabilityIEs. The granting node uses the MSH-DSCH:GrantIE to specify the range of granted minislots which selected from the free minislots reported by the requesting node. When it is sent by the requesting node, the MSH-DSCH:GrantIE acts as a grant confirmation called MSH-DSCH:ConfirmIE.

In the following, we explain the three-way handshake mechanism step by step, which is illustrated in Fig. 3.10. Note that each MSH-DSCH message transmission is just requiring one TxOpp. Therefore, it needs three time TxOpps to achieve the three-way handshake mechanism where two TxOpps are for the requesting node and

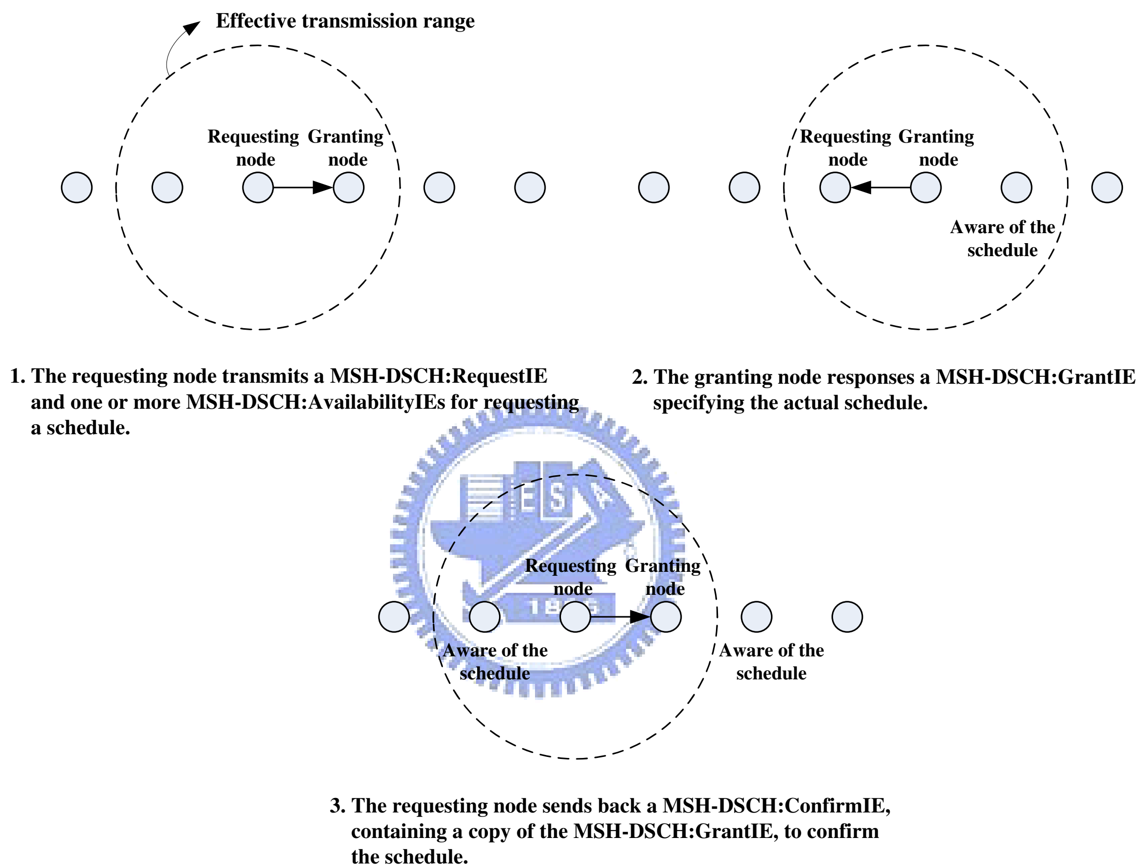


Figure 3.10: The three-way handshake mechanism to establish a schedule

one is for the granting node.

1. The requesting node first transmits a MSH-DSCH:RequestIE and one or more MSH-DSCH:AvailabilitiyIEs to ask for a schedule.

2. Upon reception of the requesting message, the granting node responds a MSH-DSCH:GrantIE specifying the actual schedule. In addition, the neighbors (except the requesting node) of the granting node shall assume that the schedule will take place as granted.

3. When receiving the granting message, the requesting node sends back a MSH-DSCH:ConfirmIE which is a copy of the MSH-DSCH:GrantIE to confirm the schedule to the granting node. Moreover, the third parties (i.e. the requesting nodes neighbors except the granting node) shall keep silence during this schedule to avoid collisions.

Chapter 4

Design and Implementation

In this chapter, we explain the design and implementation of our proposed network coding scheme for IEEE 802.16(d) mesh networks. The design of our proposed network coding scheme includes the collaboration of the MAC layer and the employed routing protocol. In our design, the packet encoding-decoding functions and coding scheduler are implemented in the MAC layer. The coding scheduler needs the next-hop information of each outgoing packet to examine whether a coding opportunity exists. Such information can be easily acquired by consulting the collaborative routing protocol, which is usually installed inside the operating system. Although our proposed network coding scheme employs a cross-layer design, the cost to integrate functions in different layers is insignificant.

We evaluate our proposed network coding scheme using the NCTUns network simulator, which can use real-life Linux protocol stack and real-life application programs to conduct simulation. In Section 4.1, we first introduce NCTUns and presents our implementations over this simulation platform. In Section 4.2, we explain the detailed formats of the network-coding packet header, which contains the information necessary to the operation of our proposed network coding scheme. We then explain the designs of the packet encoding-decoding mechanism and the coding scheduler in 4.3. Finally, in Section 4.4 we elaborate the control flow of a network coded packet from the sending node to a receiving node.

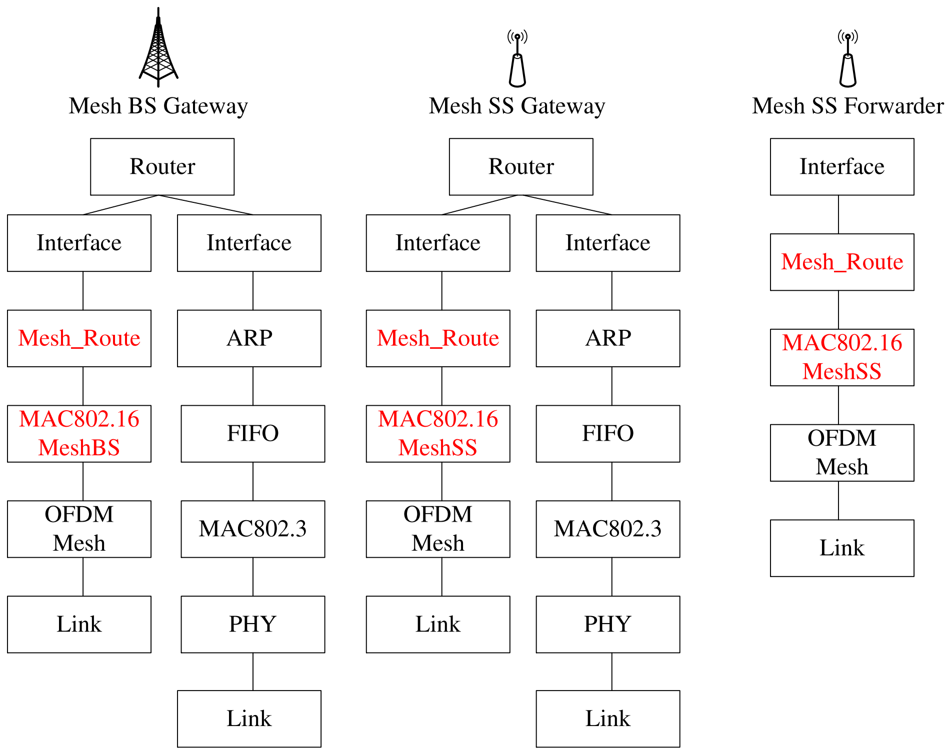


Figure 4.1: The protocol stack configuration of IEEE 802.16(d) mesh-mode modules in NCTUns

4.1 Introduction to the NCTUns Network Simulation Platform

The NCTUns network simulator [17] is a high-fidelity and extensible network simulator. By using a novel kernel re-entering simulation methodology, the protocol stacks of real-life Linux operating systems can be used to conduct simulation. As compared with other existing network simulators, real-life UNIX application programs can directly run on top of NCTUns without any modifications. This unique feature makes the simulation results generated by NCTUns more accurate and realistic.

NCTUns supports lots of different network protocols, and all of them are implemented as protocol modules. In this these, we modify the implementation of the IEEE 802.16(d) mesh-mode protocol modules, which was first developed by Chu et al. [5] and refined by Hsu et al. [7][10][19]. The architecture of the IEEE 802.16(d) mesh-mode modules is illustrated in Fig. 4.1. As shown in this figure, an IEEE 802.16(d) mesh network is composed of four modules: Mesh Route, MAC802.16 MeshBS, MAC802.16 MeshSS, and OFDM Mesh.

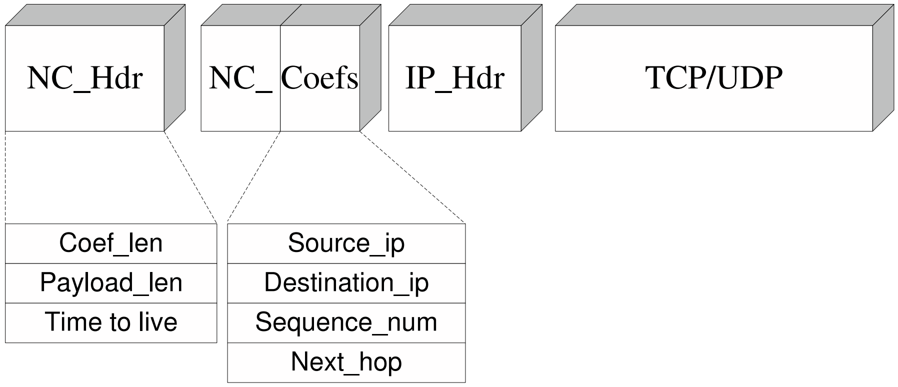


Figure 4.2: The format of a network-coded packet

The Mesh Route module determines the routes of each packet and maintains a routing table for a network node. It also implements the application programming interfaces (APIs) of the route inquiry function, which is needed by the MAC-layer coding scheduler.

On the other hand, the MAC802.16 MeshBS and MAC802.16 MeshSS modules implements the MAC-layer functions defined by the IEEE 802.16(d) mesh-mode standard. We implement the main packet coding-decoding functions in these two modules. Finally, the OFDM Mesh module implements the characteristics of OFDM channels and simulates the broadcasting nature of a wireless network.

4.2 The Format of a Network-coded Packet under Our Proposed Scheme

Our proposed network coding scheme needs to add an additional network-coding header into each outgoing packet such that intermediate nodes can differentiate whether an incoming packet is network-coded or not. Such network-coding header is inserted in front of each packet's IEEE 802.16 MAC header. The detail format of a network-coded packet is depicted in Fig. 4.2. The network-coding header can be divide into two parts: 1) the Network Coding Header (NC hdr) and 2) the Network Coding Coefficient (NC Coef). Following the network-coding header are the IP header and TCP/UDP layer-4 packet. The network-coding header is also called a shim header between the IEEE 802.16(d) MAC header and the general IP header.

In the following, we describe the fields of a network-coding header in detail.

Network Coding Header

1. Coefficient Length:

   This field indicates the number of Network Coding Coefficients. For a network-coded packet, this field is set to two because, our proposed network coding scheme only mixes packets belonging to two different flows. On the contrary, this field is set to one if the packet is fresh.

2. Payload Length:

   For a network-coded packet, its data payload corresponds to a complete IP packet. When two fresh packets of different lengths are encoded together to form a new network-coded packet, the payload length field of the new network-coded packet should be set to the length of the longer fresh packet.

3. Time-to-live:

   The Time-to-live (TTL) field shows the number of hops that a packet is allowed to be forwarded. In the next section, we will present our proposed network-encoding rules for finding coding opportunities in different coding structures. Usually, this field is set to one for all packets to avoid bandwidth wastage due to unlimited packet flooding. However, in a butterfly coding structure this field should be set to two for the operation of our proposed network coding scheme.

Network Coding Coefficient

A mixing packet may comprise two fresh packets. For each fresh packet, its Network Coding Coefficient part is composed of the following information.

1. Source IP Address:

   This field indicates the IP address of the traffic source node. In our network coding scheme, Source IP Address and Sequence Number are regarded as the unique packet ID.

2. Destination IP Address:

   This field indicates the IP address of the traffic destination node. We use this information for looking up the routing path.

3. Sequence Number:

   For every traffic source node, the Mesh Route module will assign a sequence number for each local transmitting packet. As the description above, the unique packet ID is referred to Source IP Address and Sequence Number.

4. Next Hop:

   In the network coding scheme, every node needs to overhear packets which are not for itself. For the sake of this reason, we use this information to notice the real next hop. We fill in this field by the routing information basically but sometimes modify it to perform the butterfly rule.

Adding the coding header into a packet generates a few bandwidth overheads. The Network Coding Header is of 12 bytes in length and the Network Coding Coefficient is of 16 bytes in length (for a fresh packet) or 32 bytes in length (for a mixing packet). As such, if we have a 1400-byte UDP packet, the overhead only accounts for 1.9% to 2.9%. Such a bandwidth overhead is negligible, as compared with the performance gain obtained from network coding.

4.3 Packet Encoding and Decoding

Under our proposed network coding scheme, at most two different fresh packets are exclusive-or-coded (XOR-coded) to form a network-coded packet. The reason why we use this simple coding design is explained here. According to [13], Wang at al. stated that the packet forwarding efficiency of coding two distinct packets is not

Table 4.1: The notations used in our encoding rules

Notation	Description
NH(a)	The next-hop node of packet(a)
DST(a)	The destination node of a packet(a)
TX(a)	The transmitting node of packet(a)
NBR(A)	The set of nodes that is one-hop away from node A

different from that of coding three and more distinct packets. In addition, regarding coding two distinct packets, the XOR operation can greatly reduce the computation complexity for network-coding a packet, as compared with other complicated coding operations (such as finite field operations), while, at the same time, achieve the same coding quality.

There are two advantages in our coding scheme. First, it is an easy-to-implemented and easy-to-deployed scheme. Our proposed scheme need not employ an additional protocol to find coding opportunities or coding structures. Second, as compared with COPE[12] and its extension, which only can support chain-based and star-based coding structures, our work can support chain-based, and butterfly-based coding structures at the same time. (Note that the star-based coding structure rarely happens in a real-life wireless network. As such, removing the star-based coding from our coding scheme results in insignificant performance impact.) As such, our proposed network coding scheme can find more coding opportunities that COPE-based schemes. In the following, we elaborate on the encoding mechanism employed by the proposed network coding scheme.

4.3.1 Main Encoding Rules

Chain Rule :

As shown in Fig. 4.3 (a), node B has received two packets from node A and C (denoted as Pkt(a) and Pkt(c), respectively). After receiving these two packets, node B will check whether these two packets can form a coding opportunity by checking whether the following rules are satisfied.

1. $TX(a) = NH(c)$
2. $TX(c) = NH(a)$

The chain rule is simple because only one-hop neighboring nodes (nodes A and C) of the coding node (node B) are involved. As such, the coding node can make sure that if these two conditions are satisfied, nodes A and C must possess Pkt(a) and Pkt(c), respectively. In addition, node A intends to receive Pkt(c) and node C intends to receive Pkt(a). As such, if node B transmits a network-coded packet ($Pkt(a) \oplus Pkt(c)$), node A is able to obtain Pkt(c) by XOR-ing Pkt(a) and ($Pkt(a) \oplus Pkt(c)$). On the other hand, node C is able to obtain Pkt(a) by XOR-ing Pkt(c) and ($Pkt(a) \oplus Pkt(c)$). Using the chain rule, a node can reduce the number of packet transmissions from four to three and thus increase the packet transmission efficiency by 33%.

Butterfly Rule :

The chain-based network coding is easily implemented; however, such a coding scheme can achieve at most 33% improvement of transmission efficiency. In contrast, the butterfly network coding can reduce the number of required packet transmissions from six to four, which can improve the packet transmission efficiency by 50%.

The encoding-decoding mechanisms for the chain coding structure much differ from that for the butterfly coding structure. Using the chain coding, a network-coded packet is decoded on its next-hop node. However, using the butterfly coding, the network-coded packet is decoded on the nodes that are two-hop away from the transmitting node. The operation of our butterfly coding is composed of two parts: rule checking and routing designation. The detailed design is explained below.

We use the topology shown in Fig. 4.3(b) as an example, where the network is composed of six nodes. Node A intends to deliver Pkt(a) to node F and node

C intends to transmit Pkt(c) to node D. On receiving these two packets, node B should check whether they satisfy the following rules or not.

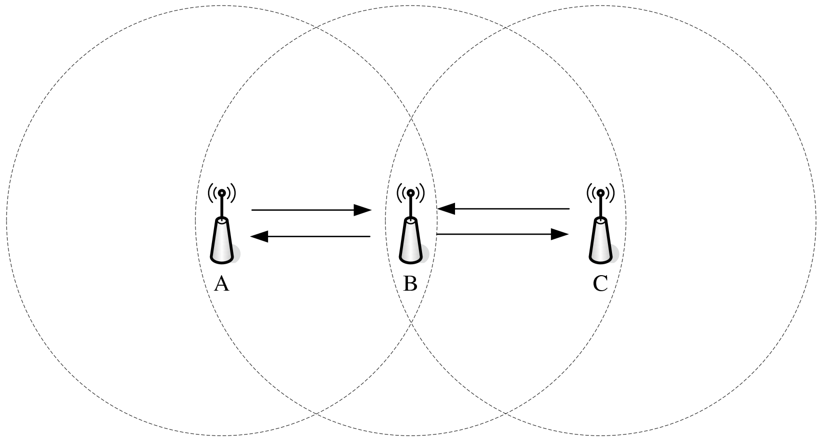
Butterfly Coding Rules

1. $NH(a) = NH(c)$
2. $DST(a) \neq DST(c)$
3. $(NBR(A) \cap NBR(E)) - B \neq \emptyset$
4. $(NBR(C) \cap NBR(E)) - B \neq \emptyset$

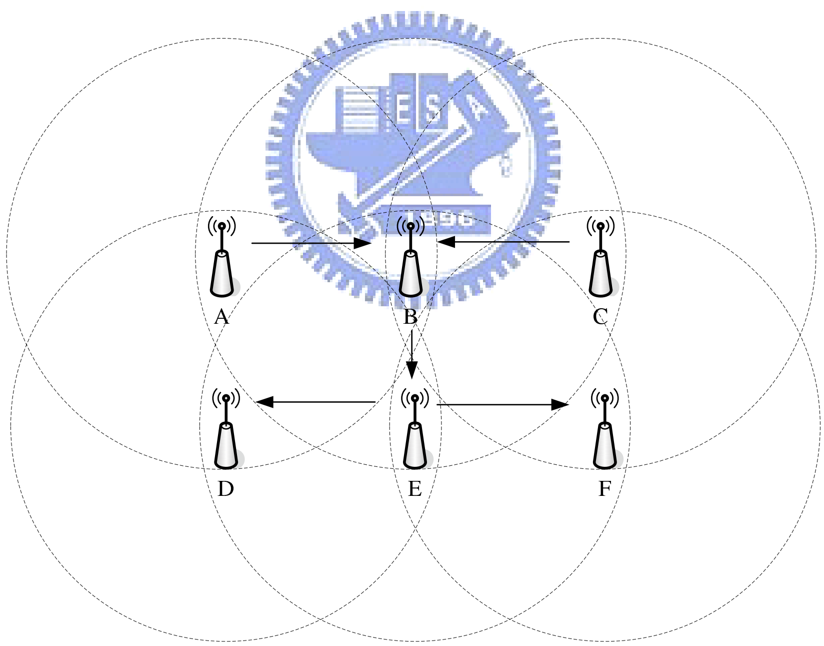
For a pair of packets, satisfying the above four rules indicates that they can form a butterfly coding opportunity. The first rule indicates that the two packets have the same next-hop node while the second rule indicates that they belong to different flows and are destined to different destination nodes, respectively.

A pair of packets satisfying the first two rules means that the routes of these two packets have a chance to perform butterfly coding. Then, the third and fourth rules check whether these two packets have opportunities to be decoded correctly in a butterfly structure, in case they are XOR-ed to form a network-coded packet. This checking is accomplished by examining if 1) nodes A and E have any common one-hop neighboring nodes other than node B and 2) nodes C and E have any common one-hop neighboring nodes other than node B. Rules (3) and (4) being satisfied means that the remedy packets for $(Pkt(A) \oplus Pkt(C))$ can reach node E' neighboring nodes, when nodes A and C transmit their own packets (i.e.,Pkt(A) and Pkt(C)) out, which means that the decoding for the network-coded packet $(Pkt(A) \oplus Pkt(C))$ can succeed on nodes D and F.

If Pkt(A) and Pkt(C) pass rules (1) to (4), node B will mix these two packets to form a network-coded packet $(Pkt(A) \oplus Pkt(C))$. Then, to make sure that Pkt(A) and Pkt(C) can reach nodes D and F, respectively, node B should designate the routes of the two mixed packets for node E. This is done by



(a) Chain rule



(b) Butterfly rule

Figure 4.3: Two encoding rules

setting the next-hop node fields of the component packets in the network coding header of the network-coded packet $(Pkt(A) \oplus Pkt(C))$. The route of Pkt(A) on node E should be set to node D and that of Pkt(C) on node E should be set to node F, respectively. In addition, node B should set the TTL field of this coded packet to 2, which indicates that node E is allowed to re-broadcast this coded packet once again.

Finally, node B can send the coded packet $(Pkt(A) \oplus Pkt(C))$ out. Upon receiving this coded packet, node E first checks whether it can decode this packet. Since it does not possess the remedy packets for this coded packet, it cannot decode this packet. In such a condition, it first decreases the TTL value of this coded packet and then checks if the TTL value is still larger than zero If so, it should re-broadcast this coded packet again. If not, it should drop this packet immediately.

Using such a design, the packet $(Pkt(A) \oplus Pkt(C))$ can be flooded to nodes D and E. Because these two node have remedy packets for $(Pkt(A) \oplus Pkt(C))$, they are able to decode the packets that they intend to receive (Pkt(C) and Pkt(A) in our example).

Our proposed scheme is feasible and easy to be implemented because, for a node, the one-hop neighboring node set of its neighboring node can easily obtained from the hello message or link state message received by the collaborative routing protocol. In addition, for an IEEE 802.16(d) mesh network, such neighborhood information can be directly obtained from control-plane messages, such as MSH-NCFG messages or MSH-DSCH messages.

Route Designation for Next-hop Node

1. $NH(a) \in ((NBR(C) \cap NBR(E)) - B)$
2. $NH(c) \in ((NBR(A) \cap NBR(E)) - B)$

4.3.2 Decoding Rules

The packet decoding mechanism of our proposed scheme is simple and explained below. Every 802.16(d) mesh-mode node must maintain a fresh packet queue to store the packets that are sent by itself or overheard from its neighbor nodes. Upon receiving an coded packet, the receiving node first looks up the fresh packet queue using the Packet ID as index (i.e, source IP address and sequence number). If one of the coded packet's component exists in the fresh packet queue, the receiving node will XOR the coded packet that it receives and the fresh component packet found in the fresh packet queue to obtain the other fresh component packet.

To save the storage space, each node periodically checks the fresh packet queue and removes packets that are obsolete. An obsolete packet is defined as a packet that has been existed in the network for more than a pre-defined time interval.

4.4 Complete Control Flow Explanation

In this section, we present the control flow of our proposed network coding scheme, which is discussed from the sending side and receiving side. The sending-side flow chart is depicted in Fig. 4.4 and the receiving-side flow chart is depicted in Fig. 4.5.

4.4.1 Sending Side

When the transmitting node sends or forwards a packet, the outgoing packet is first placed into one of the node's MAC-layer connection queue. This action then triggers the MAC layer to perform a three-way handshake procedure to create a data schedule for the connection queue, if no data schedule corresponding to the connection queue exists so far.

As shown in Fig. 4.4, after the data schedule has been created, the MAC layer first scans the connection queue and tries to find whether packets that match the chain coding rules or butterfly coding rules exist. If such packets exist, the MAC layer then 1) codes these packets; 2) removes these packets out of the connection queue; and 3) inserts the new coded packets into the connection queue. Such a scanning process is repeated until 1) the MAC layer cannot find any packets to perform network coding or 2) the total length of found packets have exceeded the length of a data burst that the MAC layer is currently allowed to transmit. After the encoding process, the MAC layer makes up a complete MAC-layer data burst for those coded packets and then transmit this data burst down to the physical layer to accomplish the sending process.

4.4.2 Receiving Side

The processing flow on the receiving side is shown in Fig. 4.5. On receiving a packet from the physical layer, the MAC layer first checks whether it is a fresh packet or not. For a coded packet, the MAC layer tries to decode it. If it cannot decode this coded packet, it then checks the TTL field and determines if it can re-broadcast it again. As aforementioned, if a packet is coded using the butterfly coding, its TTL field will be set to 2 initially. As such, if the TTL value of such a butterfly-coded packet is larger than zero, the MAC layer should insert this packet into the connection queue again to re-broadcast it.

However, in case a coded packet with TTL value being zero cannot be decoded correctly, the MAC layer should first place this packet into a coded packet queue. If the MAC layer can receive the remedy packet of this coded packet within a predefined time interval, the MAC layer can re-perform the decoding process for the coded packet. Otherwise, after the predefined time interval has elapsed, the coded packets that cannot be successfully decoded will be discarded.

On the other hand, if the MAC layer receives a fresh packet, it should first check if the packet is destined to itself or not. If it is, it should send this fresh packet to the upper layer. If not, it should store a copy of this packet in the fresh packet queue for future possible decoding. (This is because such a packet may be the remedy packet of future incoming network-coded packet.)

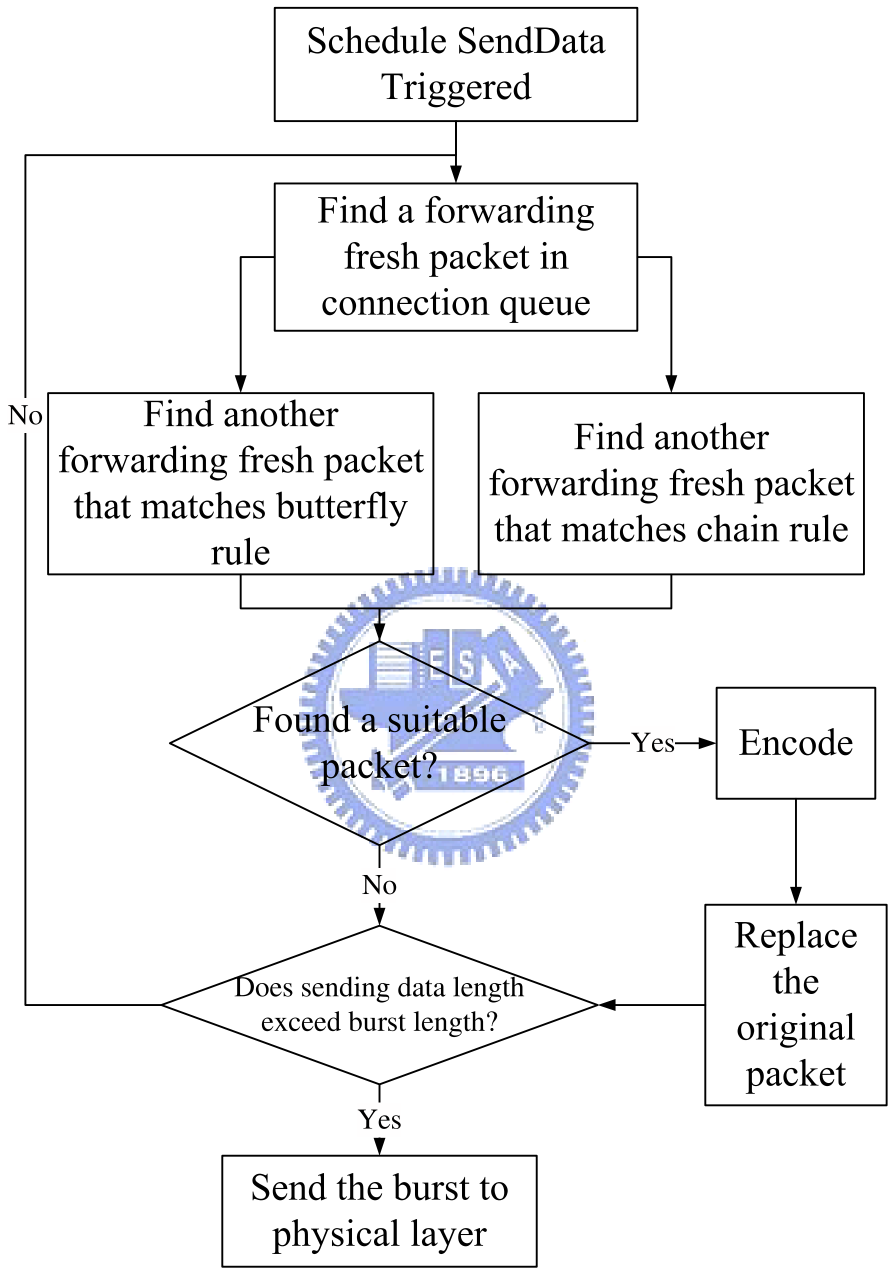


Figure 4.4: The sending flow chart

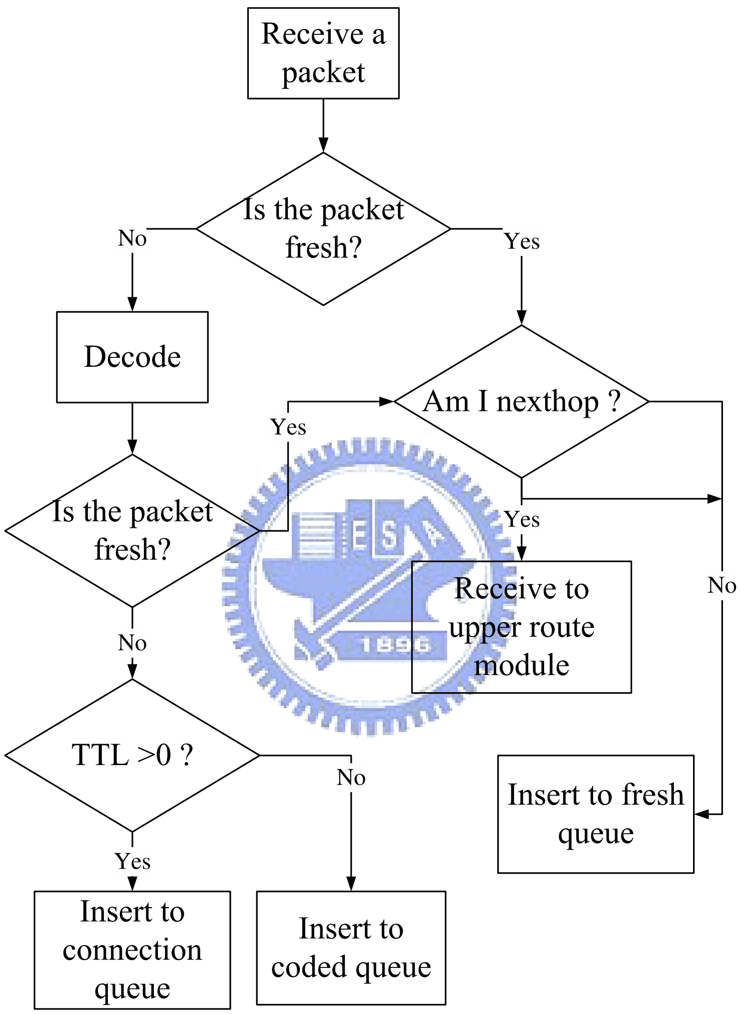


Figure 4.5: The receiving flow chart

Chapter 5

Four-way Handshake Procedure

In this chapter, we discuss the impacts caused by the hidden terminal problem and the extended hidden terminal problem on a network coding scheme. Such an issue is rarely addressed in the previous studies, which greatly weakens the practicability of network coding over a real-life wireless network. To design a practical network coding scheme, in this chapter we explain the problems caused by hidden terminal nodes and extended hidden terminal nodes and propose a solution that is feasible in IEEE 802.16(d) mesh network.

The remainder of this chapter is organized as follows. We first discuss why the packet collision problem can affect the performance of a network coding scheme in Section 5.1. Then, we point out the extended hidden terminal problem that frequently occurs in a network-coded network and explain why it greatly degrades network coding performances in Section 5.2. In Section 5.3, we present our four-way handshake design to solve this problem in detail. Finally, we propose an algorithm to determine the activation time of a data schedule to further reduce the potential bandwidth wastage that may result from our proposed four-way handshake design in Section 5.4.

5.1 Packet Collision Problem in a Wireless Network-coded Network

It has been known that, in a wireless network, the hidden terminal problem can result in frequent packet collisions. In a traditional IEEE 802.11 network, the RTS-CTS mechanism is used to prevent transmitted data packets from being collided. However, the RTS-CTS mechanism also greatly increases the bandwidth and time cost for a packet transmission. (Using the RTS-CTS mechanism, a data packet transmission should be accompanied by two control packet, i.e., the RTS and CTS packets.) As such, it significantly reduces the throughput performance of a wireless network.

In [12][8][11], Katti et al. have pointed out that the hidden terminal problem can degrade the performance of a network coding scheme due to the packet collisions that it causes. In addition, they also point out that using the RTS-CTS mechanism to solve the hidden terminal problem will diminish the performance gain achieved by network coding. Due to this reason, most of the previous network coding research assume that a wireless network is collision-free and no hidden terminals (and extended hidden terminals) are present. However, such an assumption is unrealistic for a real-life wireless network.

As explained in Section 3.2.5, the IEEE 802.16(d) mesh mode uses a reservation-based approach to schedule data transmission and employs a three-way handshake procedure to eliminate hidden terminals. Because a three-way handshake procedure can reserve a data schedule that spans at most 1.28 second, the bandwidth and time overheads for avoiding hidden terminals in 802.16(d) mesh networks can be amortized and less significant, as compared with those in 802.11 networks.

However, the three-way handshake procedure designed by the 802.16(d) mesh mode does not consider the extended hidden terminal problem that frequently occurs in a wireless network using network coding. The reasons why an extended hidden terminal problem occurs are explained here.

In Fig. 5.1, the circles centered at nodes B and D represent the available trans-

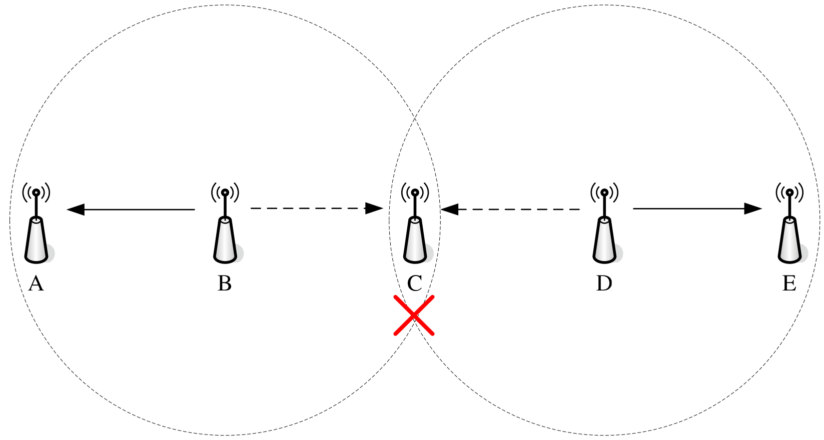


Figure 5.1: The extended hidden terminal problem in chain networks

mission range of these two nodes, respectively. The solid arrow denotes the direction on which a node intends to its packets and the dotted arrow denotes the direction on which its neighboring nodes is likely to overhear this packet. Although the three-way handshake mechanism of the 802.16(d) standard can protect the transmitted packets from being collided on the intended receiver (nodes A and E in this example case), these packets are still collided on node C, which should overhear these packets to perform network coding.

In a traditional routing-based wireless network, such packet collisions are allowed and do not reduce the effective throughputs experienced by intended receiving nodes. Nevertheless, in a network-coded wireless network, if network coding is performed, coded packets may be collided on nodes that should overhear them for decoding network-coded packets. For example, in the example show in Fig. 5.1, node D should transmit a coded packet to both node A and node C to increase transmission efficiency. However, using the three-way handshake mechanism, this coded packet are protected only on node A. In contrast, it is collided with the one transmitted by node D on node C. As such, this packet cannot be correctly decoded on node C, which greatly degrades the performance gain achieved by network coding.

Such an extended hidden terminals can also appear in butterfly coding structures. As shown in Fig. 5.2, in a butterfly-based network-coded network, nodes D and F are very likely to lose the remedy packets required to decode network-coded packets broadcasted by node E, due to the collisions caused by extended hidden terminals. (In this case, the node pairs (A,E) and (C,I) are extended hidden terminals to

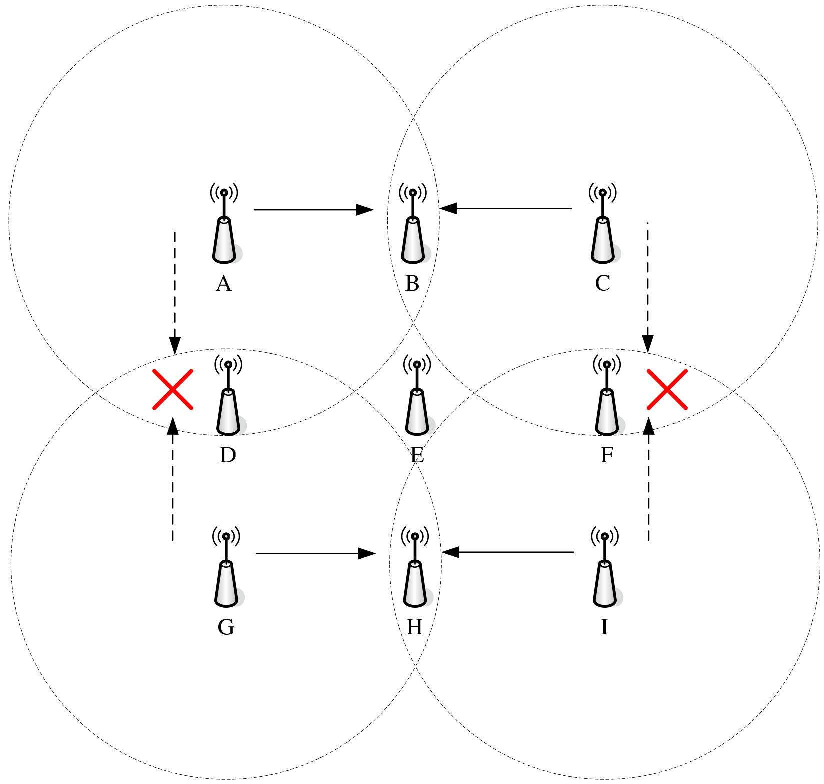


Figure 5.2: The extended hidden terminal problem in butterfly networks

each, respectively.) Note that, because the original three-way handshake mechanism cannot solve this sort of hidden terminals. Hence, we call this sort of hidden terminal problems as the "extended hidden terminal problem."

5.2 Modified Control Messages

To solve the "extended hidden terminal problem", we enhance the original WiMAX three-way handshake scheduling procedure and name it to "four-way handshake" mechanism. Before describing the details of the "four-way handshake" mechanism, we shall explain the control message formats which we modified for the new scheduling procedure.

In Section 3.2.5, we express the original three-way handshake and the control message the nodes exchanged. Similarly, the four-way handshake mechanism uses the same MSH-DSCH control messages to notify the neighbors. However, we make a little modification of the message formats. As depicted in Fig. 5.3, we use the reserved field to achieve our goals. The two reserved bits of the MSH-DSCH message should be all zero in standard. We now take one bit to indicate this MSH-DSCH is using the four-way handshake procedure.

In addition, we create a new IE named "extended confirm". This new IE copies all fields in Grant IE and adds additional 5 bits for extended usage. First, we create a 1-bit ExtConfirm field to distinguish from other kinds of Grant IEs. In the standard, the Grant IE and Confirm IE are the same except the direction bit. Therefore, the new ExtConfirm field also acts the notification bit in our design. Second, we add a Frame Prefix field where its length is 4-bits. The Start Frame Number appending to the Frame Prefix forms the integral Frame Number. In the IEEE 802.16 standard, the total length of a Frame Number is 12 bits. For the sake of saving control message length, the Grant IE only picks last 8-bits Frame Number to transmit and receivers need to recover the actual value by the corresponding Availability IEs or Grant IEs. Unfortunately, the design in IEEE 802.16 standard does not consider the situations in the four-way handshake procedure. That is, we do not have enough information to recover the real Frame Number in the ExtConfirm IEs. Therefore, we add the Frame Prefix back to insure the correctness of Frame Number in the ExtConfirm IEs. The format comparison is shown as in Fig. 5.4.

In a word, if we receive a MSH-DSCH message whose Reserved field is 00, we recognize the length of the Grant IEs in this MSH-DSCH is 40-bits. On the other hand, if the Reserved field of a MSH-DSCH message is 01, we recognize the length of the Grant IEs in this MSH-DSCH is 45-bits. The max number of Grant IEs in a MSH-DSCH message is also changed because we modified the Grant IE formats. In the IEEE 802.16 standard, The max number of Grant IEs is up to 63 in a MSH-DSCH message (i.e total length is 2520 bits). In our four-way handshake procedure, the max number of Grant IEs is set to 56. However, the number of Grant IEs in a MSH-DSCH message rarely exceeds the limit in the simulations.

(Original MSH_DSCH message format)

8 bits	1 bit	1 bit	6 bits	4 bits	4 bits	6 bits	2 bit	variable bits
Msg Type =41	Coordination Flag	Grant/Req Flag	Sequence count	No. Req	No. Avail	No. Grant	Reserved =00	IEs

(Modified MSH_DSCH message format)

8 bits	1 bit	1 bit	6 bits	4 bits	4 bits	6 bits	2 bit	variable bits
Msg Type =41	Coordination Flag	Grant/Req Flag	Sequence count	No. Req	No. Avail	No. Grant	Reserved =01	IEs

Figure 5.3: The format of the modified MSH-DSCH message

(Original Grant IE format)

8 bits	8 bits	8 bits	8 bits	1 bit	3 bits	4 bits
Link ID	Start Frame Number	Minislot Start	Minislot Range	Direction	Persistence	Channel

(Modified Grant IE format)

8 bits	4 bits	8 bits	8 bits	8 bits	1 bit	3 bits	4 bits	1 bit
Link ID	Frame Prefix	Start Frame Number	Minislot Start	Minislot Range	Direction	Persistence	Channel	ExtConfirm

Figure 5.4: The format of the modified Grant IE message

5.3 Four-way Handshake Procedure Design

In the following, the "four-way handshake" procedure is presented. The four-way handshake procedure extends from the original three-way handshake mechanism in the IEEE 802.16 standard. Fig. 5.5 illustrates the procedure of the four-way handshake procedure. The solid arrows are represented the real direction the node want to send packets and the dotted arrows mean the overheard direction, respectively. At the beginning, the requesting node sends a Request IE and the granting node responses a Grant IE. Then the requesting node transmits a Confirm IE to accomplish the original three-way handshake procedure. In the previous section, Fig. 3.10 shows that the aware areas and the transmission protection between the requesting node and the granting node in the three-way handshake procedure. Now we extends the fourth way at the neighbors of the requesting node. After receiving a Confirm IE which is not for themselves, these nodes will copy the Confirm IE to an Extended Confirm IE and transmit to their neighbors (i.e the two-hop neighbors of the requesting node). This mechanism makes sure that the the requesting node

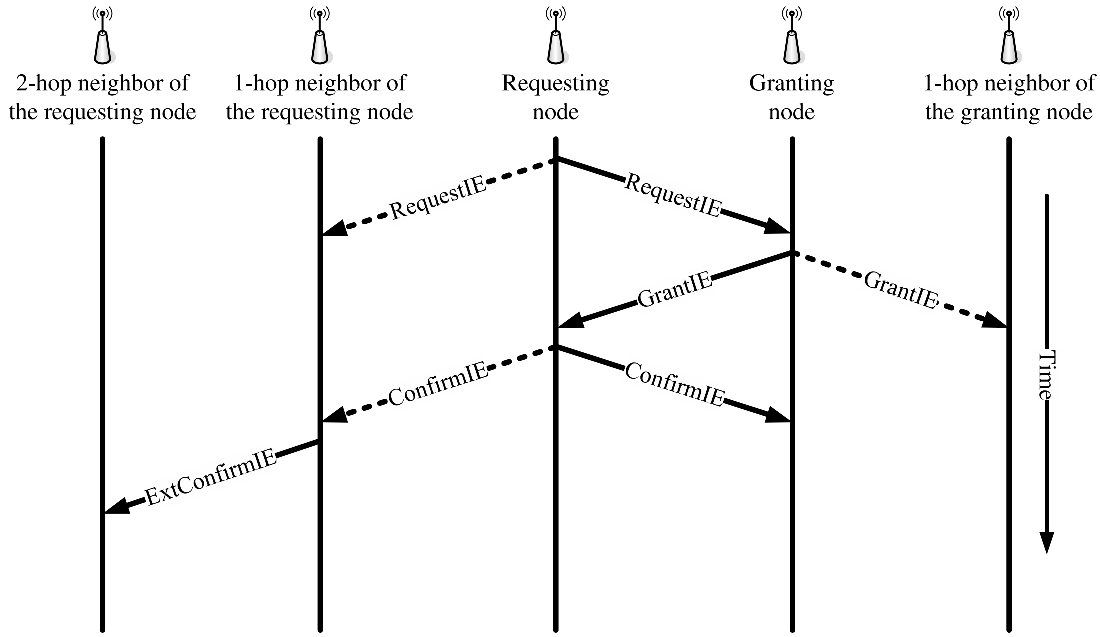


Figure 5.5: The four-way handshake procedure

and its two-hop neighbors share the same scheduling information and they will not transmit packets at the same time.

In the original design, if the mini-slots of an extended confirm overlap with the existing schedule mini-slots, the scheduler may reduce the granting frame ranges to prevent from collisions. Take Fig. 5.6 (a) for example, based on the IEEE 802.16(d) standard, if the original 128-frame schedule is overlapped by another schedule (in this case, it means the extended confirm), the scheduler will try to use multiple 32-frame schedules to replace. This original process wastes some mini-slots and may reduce the performance. On the contrary, we enhance the handling function of extended confirms. As shown in Fig. 5.6 (b), the scheduler also use the original 128-frame schedule to grant. At the actual transmission time, each node (which overlaps the mini-slots with other nodes) runs an algorithm to decide what node can send data within the "Overlapped Mini-slots". If some mini-slots in an overlapped frame are not really overlapped with other nodes (see the first four frames in Fig. 5.6), the node can still transmit packets at that time.

In the following, we list the decision algorithm in Algorithm 1. This algorithm is quite simple and distributed. However, we do not consider the fairness problem in this algorithm and we also discuss this issue in Chapter 7.

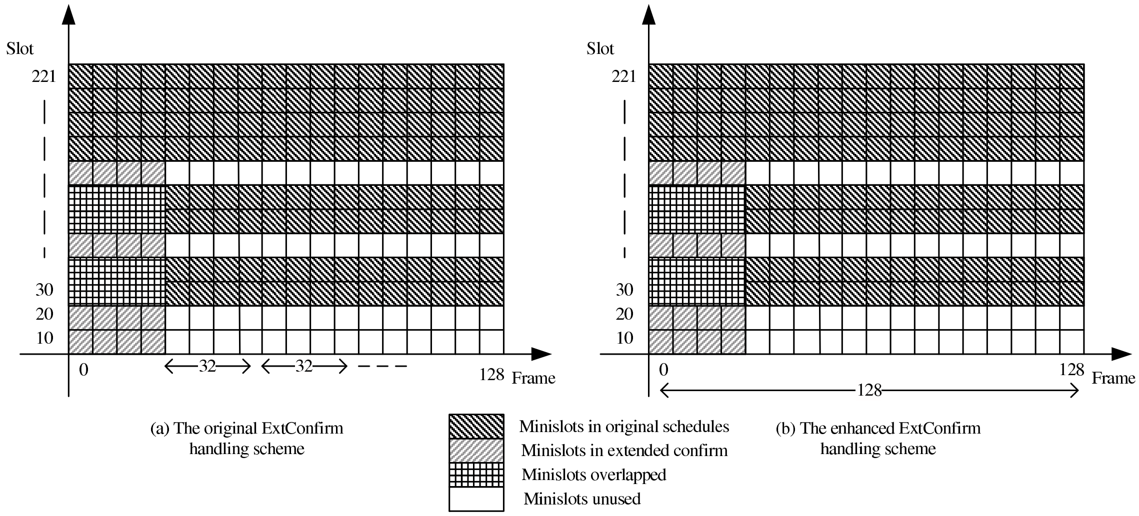


Figure 5.6: The mini-slot usages when dealing with the extended confirm

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Algorithm 1 The Transmitting Node Decision Algorithm in Enhanced Extended Confirm Handling Procedure

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$ExtCfm \Leftarrow the\ overlapped\ extended\ confirm\ schedule$

$OriSch \Leftarrow the\ overlapped\ original\ schedule$

if $ExtCfm \rightarrow startframe > OriSch \rightarrow startframe$ **then**

  $OriSch\ wins\ the\ overlapped\ minislots$

else if $(ExtCfm \rightarrow startframe = OriSch \rightarrow startframe)$ and $(ExtCfm \rightarrow startminislot > OriSch \rightarrow startminislot)$ **then**

  $OriSch\ wins\ the\ overlapped\ minislots$

else if $(ExtCfm \rightarrow startframe = OriSch \rightarrow startframe)$ and $(ExtCfm \rightarrow startminislot = OriSch \rightarrow startminislot)$ and $(ExtCfm \rightarrow nodeID > OriSch \rightarrow nodeID)$ **then**

  $OriSch\ wins\ the\ overlapped\ minislots$

else

  $ExtCfm\ wins\ the\ overlapped\ minislots$

end if

5.4 Data Schedule Holdoff Algorithm

The four-way handshake procedure works successfully bases on a hypothesis. That is, the data schedule time of the requesting node needs to start after all the two-hop neighbors receive the Extended Confirm IEs. If the data schedule starts earlier, collisions may still occur due to the missed Extended Confirm IEs. To solve this problem, we design an algorithm to compute the suitable data schedule starting time. As described in Section 3.2.4, the WiMAX nodes exchange MSH-NCFG messages with their neighbors periodically and the next transmitting time information is involving in the MSH-NCFG messages. Therefore, we can consider all the corresponding nodes and choose a time late enough to wait the Extended Confirm sending out.

The data schedule holdoff algorithm is explained in Algorithm 2. We divide this problem into two cases. First, if the Next Txopps of the granting node is before the Next Txopps of the requesting node itself, the requesting node tries to look up all the Next Txopps of its one-hop neighbors. The greatest one refers to the last time extended confirm will be sent out. Therefore, the requesting node will start the data schedule after the greatest Txopps. However, the second case is unpredictable. If the Next Txopps of the granting node is before the the requesting node, the requesting node can not understand when the granting node will response the Grant IE. Consequently, the requesting node choose the original data schedule starting time without any back-off.

Algorithm 2 The Data Schedule Holdoff Algorithm

$NBR(x)\ :\ one\ hop\ neighbors\ of\ node_x$

$G \Leftarrow the\ granting\ node$

$MyNextTxopp \Leftarrow the\ transmission\ time\ of\ my\ next\ control\ message$

$NbrList \Leftarrow (NBR(myself) - NBR(G) - G)$

$MaxTxopp \Leftarrow MyNextTxopp$

if $The\ next\ txopp\ of\ G \leq MyNextTxopp$ **then**

    for $node_i \in NbrList$ **do**

        $txopp_i \Leftarrow the\ max\ next\ txopp\ of\ node_i$

        if $txopp_i > MaxTxopp$ **then**

            $MaxTxopp = txopp_i$

        end if

    end for

    $return\ MaxTxopp$

else

    $return\ MyNextTxopp$

end if

Chapter 6

Performance Evaluation

In this chapter, we use the NCTUns network simulator[17] to evaluate the performances of our network coding scheme. In the following, we first describe the parameters used in the simulations in Section 6.1. In Section 6.3 and Section 6.2, we then show the different environments and measured metrics in the simulations. Subsequently, the simulation results compared with the original IEEE 802.16 standard are presented in Section 6.4. Finally, in Section 6.5, we discuss the system resource usages in the network coding scheme and display that this scheme can be used in the real-world devices.

6.1 Simulation Parameters Description

- Connection Queue Size

  Different connection queue sizes in a WiMAX node represent the maximum capacities of the incoming traffic flows. We simulate the connection queue sizes from 50 packets to 1000 packets to study the influence of performance about the connection queue size.

- Fresh Queuing Period

  The fresh queuing period is a duration storing the fresh packets. If the fresh queuing period is too small, the encoding packets may not find the remedy packets to decode due to the flush of the fresh queue. We compare different fresh queuing periods to study the performance effects.

6.2 Simulation Metrics

Our performance evaluation observes the following metrics.

- Aggregate Average Traffic Flow Throughput (AATFT)

  We use the aggregate average application throughput of all traffic flows to show the network capacities in the simulations. The Aggregate Average Traffic Flow Throughput (AATFT) is defined as follows:

$$AATFT = \sum_{i=1}^{N} t_i \tag{6.1}$$

  where t_i is the average throughput of the ith traffic flow and N denotes the total number of traffic flows in the simulation.

- Average End-to-end Delay (AETED)

  We compute the application-to-application delay of every UDP traffic flow. The Average End-to-end Delay (AETED) is defined as follows:

$$AETED = \frac{\sum_{i=1}^{N} AD_i}{N} \tag{6.2}$$

  where AD_i denotes the average delay of the ith traffic flow and N denotes the total number of traffic flows in the simulation.

- Reduced Ratio of Collisions (RRC)

  As mentioned in Chapter 5, we proposed the four-way handshake scheme to eliminate the EHT problem. The Reduced Ratio of Collisions (RRC) is the ratio of the measured collision numbers in the physical layer (i.e OFDM_MESH module) with and without four-way handshake scheme.

$$RRC = \frac{Collisions\ with\ FourWayHandshake}{Collisions\ without\ FourWayHandshake} \tag{6.3}$$

6.3 Simulation Environment Description

In this section, we explain the simulation topologies and traffic patterns. We use the aggregate throughput of all traffic flows to show the network utilization in the simulations. All the traffic flows start after the IEEE 802.16 Network Initialization Processes (as in Section 3.2.2, Section 3.2.3) and continue for 50 seconds. The traffic flows have Poisson arrivals and fixed 1400-byte payload size.

In the following, the simulation environments are depicted.

- Chain Topology

  As shown in Fig. 6.1 (a), the basic chain network is also called the Alice-and-Bob network. We set two flows, one is from node 1 to node 3 and the other go through the reverse direction. Each flow generates packets every 0.001 second averagely.

- Butterfly Topology

  As shown in Fig. 6.1 (b), there are also two flows in the butterfly network. We set one from node 1 to node 6 and the other from node 3 to node 4. Each flow generates packets every 0.002 second averagely.

- 7-node Chain Topology

  As shown in Fig. 6.2, there are also two flows in this network. We set one from node 1 to node 7 and the other from node 7 to node 1. Each flow generates packets every 0.001 second averagely.

- 9-node Grid Topology

  As shown in Fig. 6.3, this is a case of a lot of EHT problem occurrences as we explained in Section 5.1. We set four flows in the this network, which are $1 \rightarrow 6$, $3 \rightarrow 4$, $9 \rightarrow 4$ and $7 \rightarrow 6$. Each flow generates packets every 0.001 second averagely.

- 25-node Grid Topology

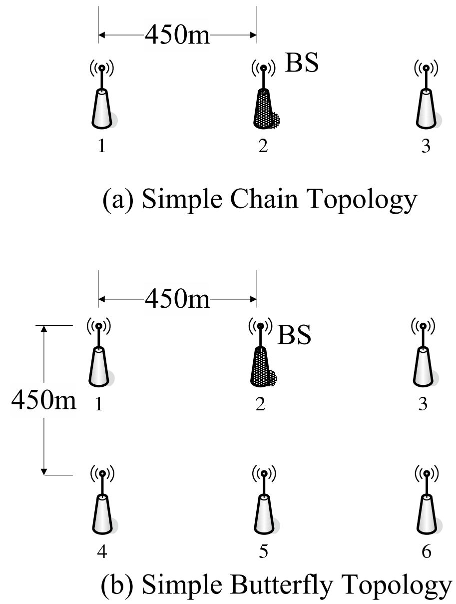

Figure 6.1: The two simple topologies

Figure 6.2: The 7-nodes chain topology

As shown in Fig. 6.4, this large topology contains 16 flows. Each flow generates packets every 0.005 second averagely. The following eight pair of nodes are acting the source node and destination node respectively: (2,14), (3,25), (4,24), (5,23), (6,22), (7,21), (12,16) and (11,17).

6.4 Simulation Results

In this section, we present the simulation results under different cases.

6.4.1 Network Coding Throughput Improvement

In Fig. 6.5, Fig. 6.6, Fig. 6.11, Fig. 6.10 and Fig. 6.12, our network coding scheme uplifts the throughput under different network topologies. Our network coding scheme

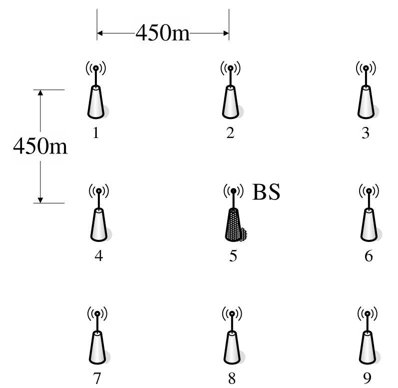

Figure 6.3: The 9-nodes grid topology

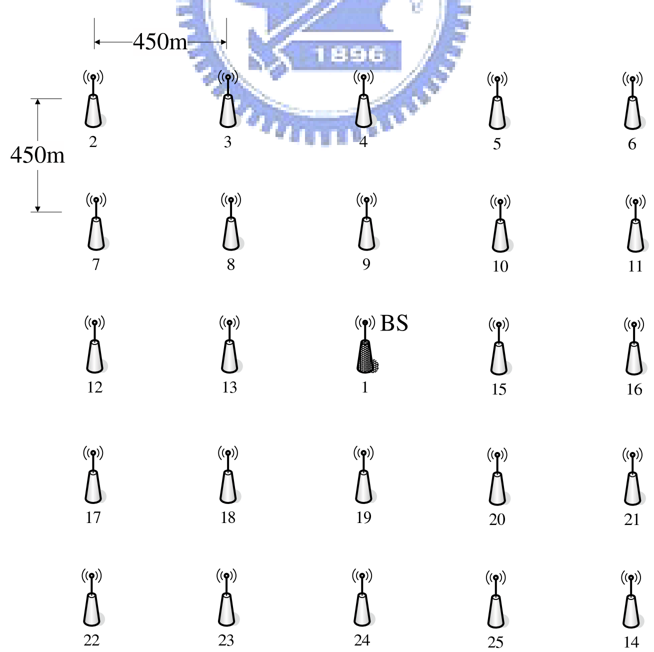

Figure 6.4: The 25-nodes grid topology

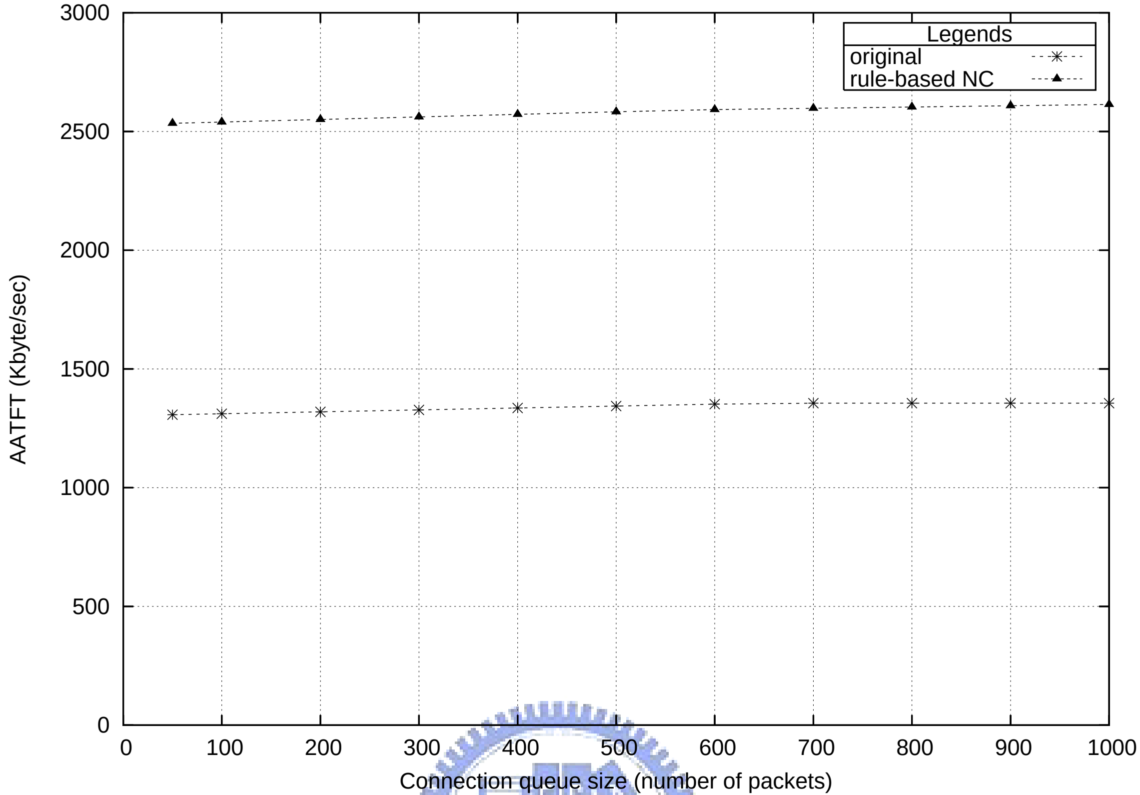

Figure 6.5: The throughput of the Alice-and-Bob network

gains a 89% improvement as shown in Fig. 6.5 and 61% improvement as shown in Fig. 6.6. These two simple topologies valid the two encoding rules as mentioned in Section 4.3.1. In addition, in the more complicated cases, our network coding scheme also increases the throughput from 24% to 57%. The results are depicted as in Fig. 6.11, Fig. 6.10 and Fig. 6.12.

6.4.2 Under Different Fresh Clear Timers

As mentioned in Chapter 4, we clear the fresh packet queue periodically in order to save the storage space. The timing to trigger the cleaning process is considered as the fresh clear time. In the network coding scheme, we need the queued fresh packets to decode a coded-packet. If the fresh clear time is too small, an useful fresh packet may be regarded as an obsolete one and dropped. The performance will decrease duo to the fail-decoded mixing packets. We show this situation by

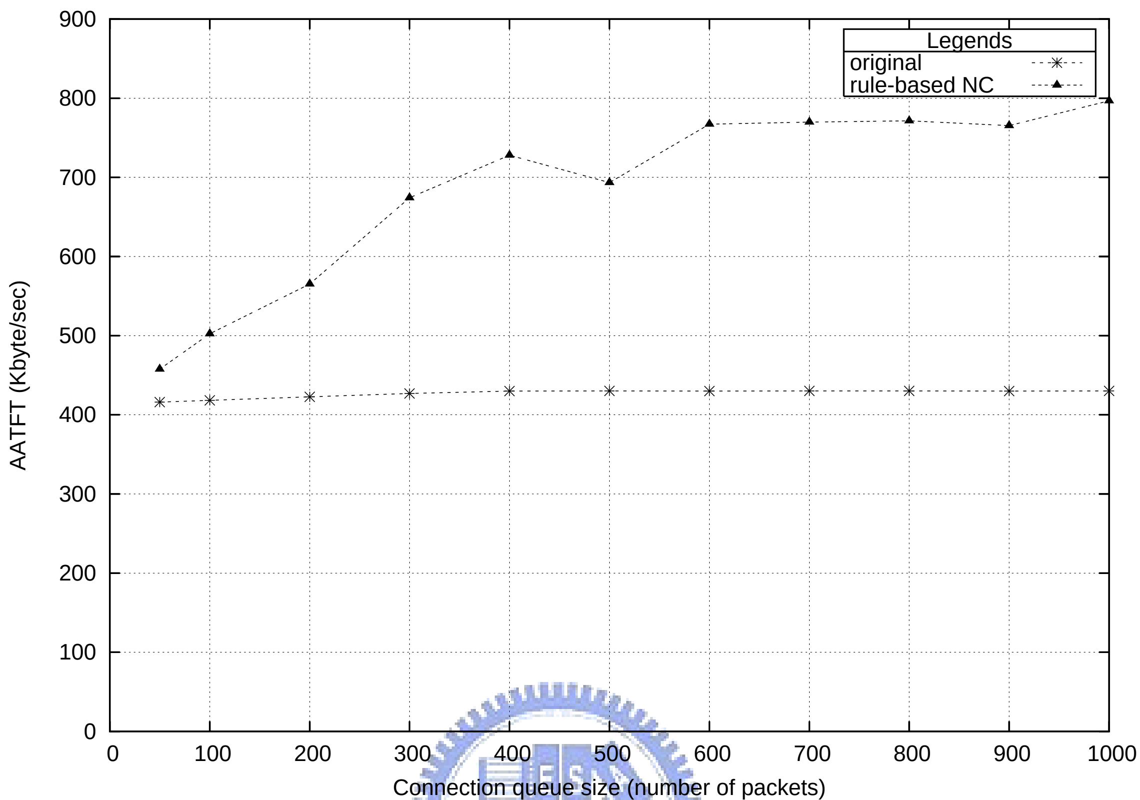

Figure 6.6: The throughput of the butterfly network

Table 6.1: Average end-to-end delay under different connection queue sizes in Alice-and-Bob networks

connection queue size (packets)	original		rule-based nc	
	AETED (sec)	Sdv	AETED (sec)	Sdv
50	0.1169	0.0005	0.0470	0.0041
100	0.2196	0.0005	0.0504	0.0073
200	0.4235	0.0005	0.0504	0.0073
300	0.6257	0.0004	0.0504	0.0073
400	0.8262	0.0004	0.0504	0.0073
500	1.0249	0.0003	0.0504	0.0073
600	1.2220	0.0003	0.0504	0.0073
700	1.4175	0.0002	0.0504	0.0073
800	1.6114	0.0002	0.0504	0.0073
900	1.8037	0.0001	0.0504	0.0073
1000	1.9944	0.0001	0.0504	0.0073

Table 6.2: Average end-to-end delay under different connection queue sizes in butterfly networks

connection queue size (packets)	original		rule-based nc	
	AETED (sec)	Sdv	AETED (sec)	Sdv
50	0.204	0.004	0.210	0.013
100	0.367	0.000	0.343	0.032
200	0.682	0.000	0.562	0.050
300	0.997	0.001	0.727	0.010
400	1.414	0.089	0.911	0.051
500	1.623	0.011	1.003	0.211
600	1.930	0.013	1.303	0.051
700	2.235	0.016	1.703	0.190
800	2.620	0.030	1.686	0.138
900	2.850	0.006	1.651	0.527
1000	3.157	0.011	1.802	0.678

three environments: the 6-node butterfly network, the 7-node chain network, and the 9-node grid network.

Fig. 6.7, Fig. 6.8 and Fig. 6.9 reveal the AATFT under different fresh clear timers. We can observe that when connection queue size increases, the AATFT with small fresh clear time is decreasing. We explain the reason in the following. If the connection queue is large, the incoming packets will be stored in connection queue for a while. However, when it is finally encoded and drained out, the destination node eliminates the remedy packets due to the small fresh clear time.

6.4.3 Under Four-Way Handshake Procedure

In Chapter 5, we elaborated the extended hidden terminal (EHT) problem and proposed our solution: the four-way handshake procedure. In the following, we first present the Reduced Ratio of Collisions (RRC) of three different topologies and show the great reductions in our scheme. Second, we display the throughput results by AATFT.

Reduced Ratio of Collisions:

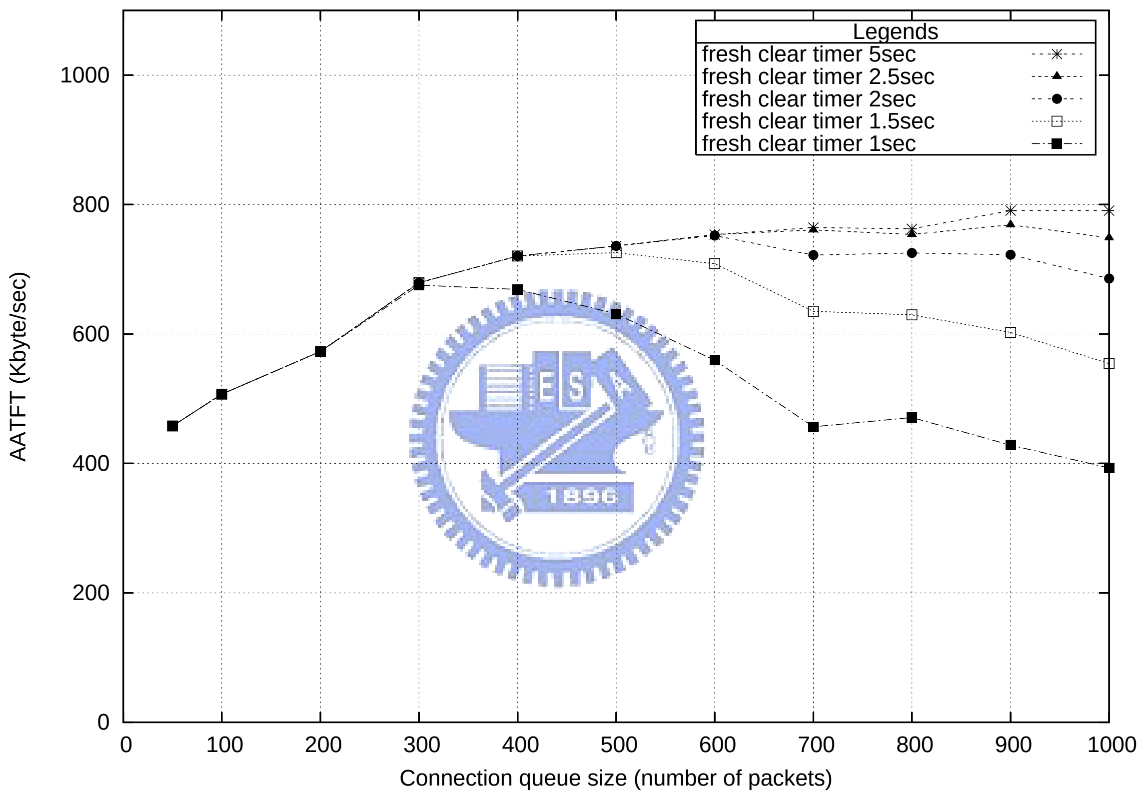

Figure 6.7: The throughput of butterfly network under different fresh clear timers

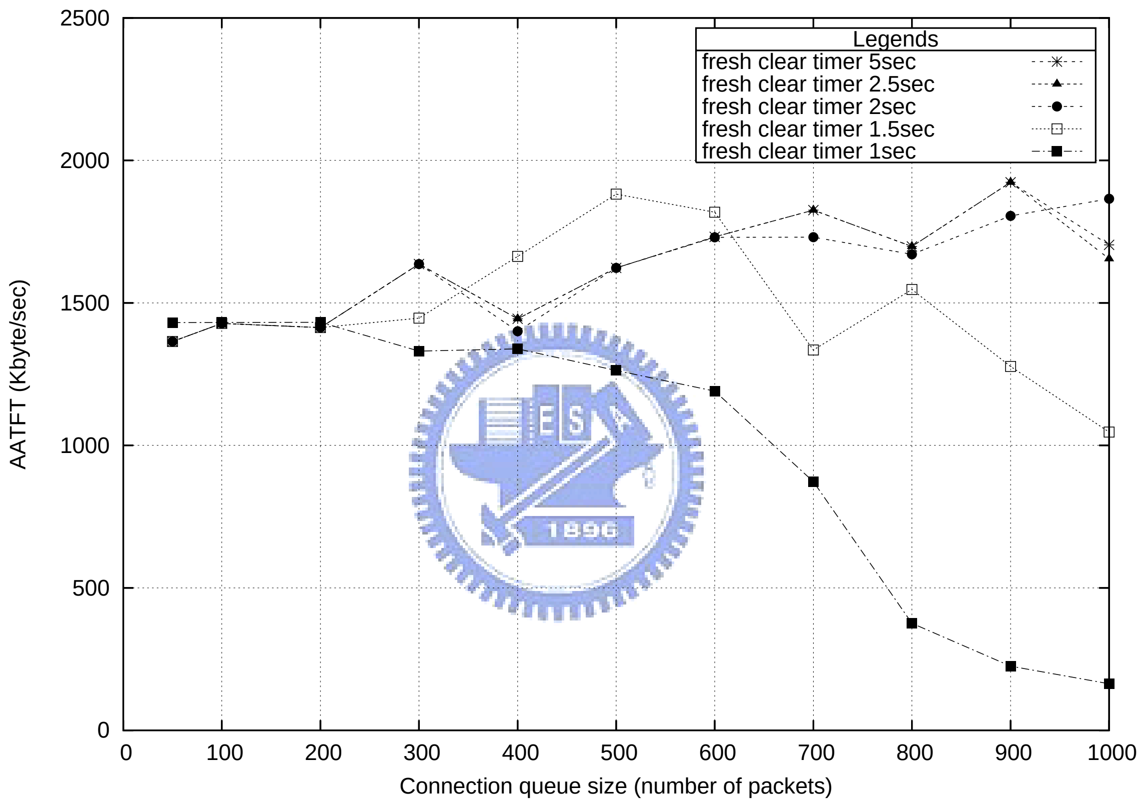

Figure 6.8: The throughput of 7-node chain network under different fresh clear timers

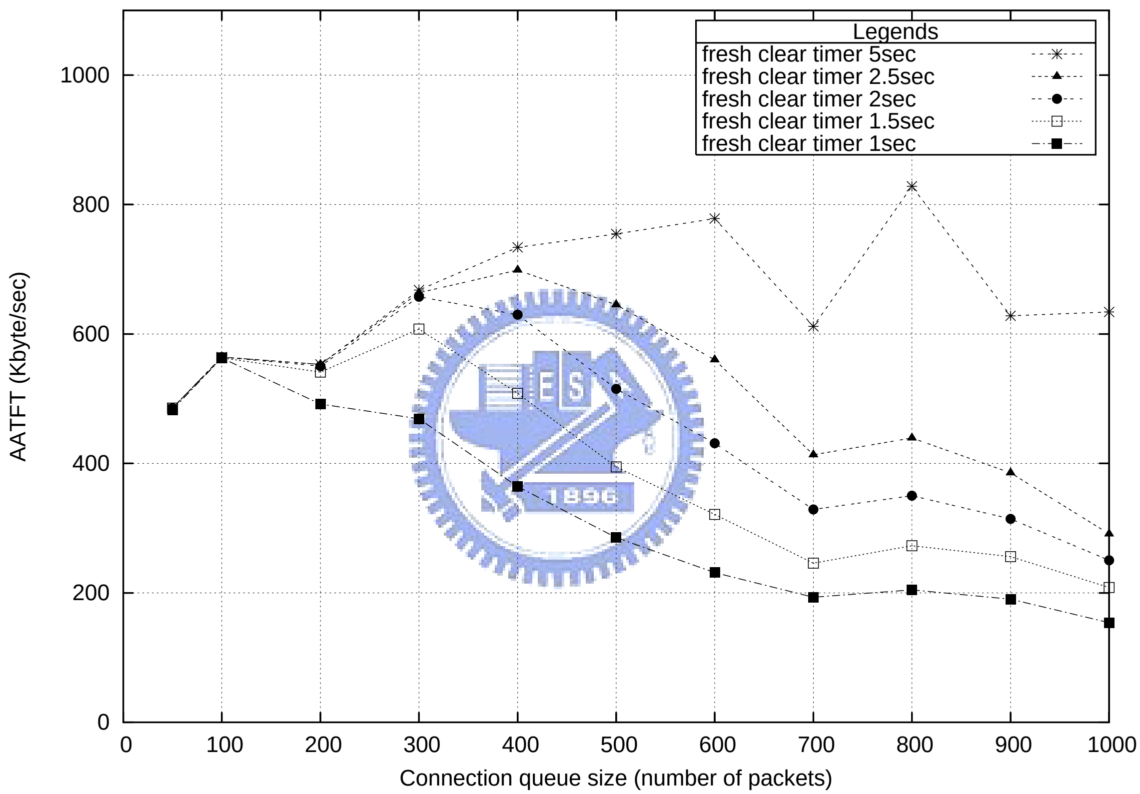

Figure 6.9: The throughput of 9-node grid network under different fresh clear timers

Table 6.3: The average reduced ratio of collisions in the four-way handshake procedure

	Without Data Schedule Holdoff Algorithm	With Data Schedule Holdoff Algorithm
7-node chain network	29.86%	17.5%
9-node grid network	1.49%	0.02%
25-node grid network	1.82%	1.16%

The RRC denotes the ratio of collision preventions in our four-way handshake procedure. We compare the four-way handshake procedure with and without the data schedule holdoff algorithm proposed in Section 5.4. The results are shown in Table 6.3. Obviously, our four-way handshake procedure highly reduces the collision occurrences and the Data Schedule Holdoff Algorithm is more effective than the original four-way handshake scheme.

Aggregate Average Traffic Flow Throughput:

After discussing the reduction of collision occurrences, we then show the throughput results comparing with the original WiMAX, network coding scheme, four-way handshake procedure and four-way handshake with data schedule holdoff. Fig. 6.10 shows the AATFT values versus connection queue size under different schemes in the 7-node chain network. The original WiMAX throughputs are much lower than the other network coding schemes. In addition, we have better results when using four-way handshake scheme. This result indicates the collision damage is solved by our design.

In Fig. 6.11, the EHT problem frequently occurs in such a network topology. We can find out that the throughput with network coding is sometimes close to the original WiMAX scheme. This result shows that the collision actually affects the goodput generated by network coding scheme because the throughput highly increases when using the four-way handshake procedure to prevent the collision occurrences. The result in Fig. 6.12 also displays the improvements by using the

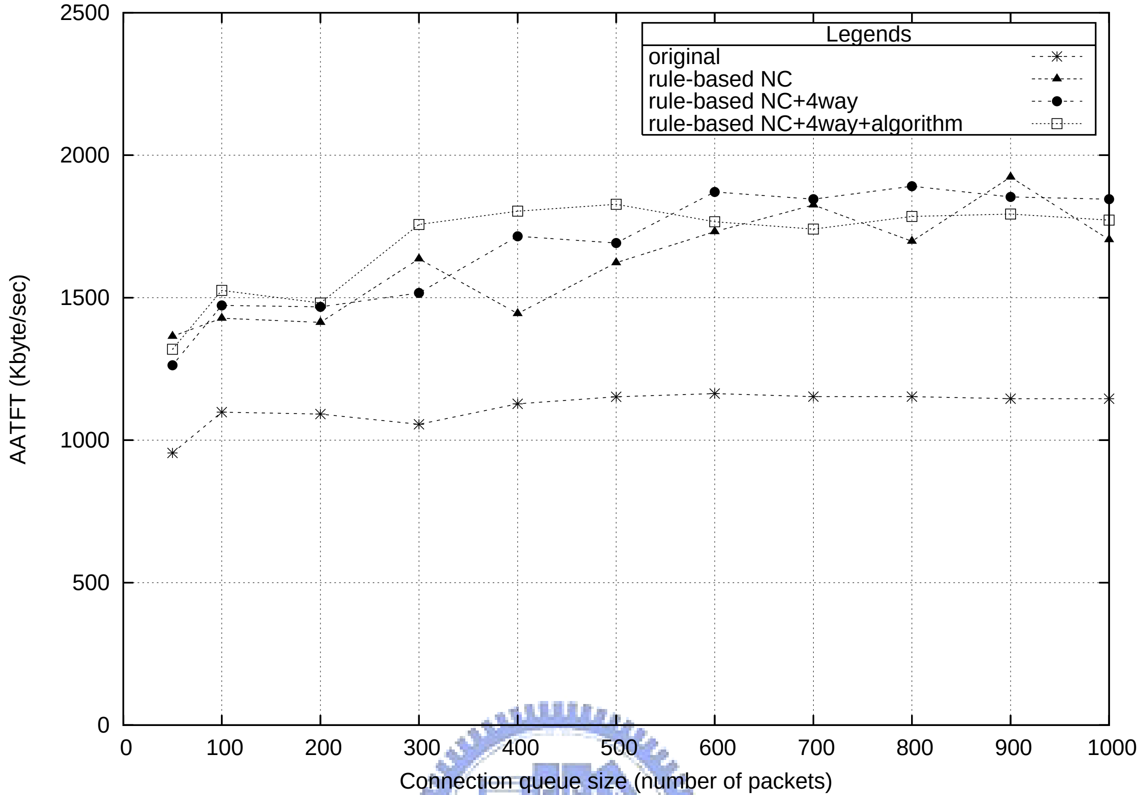

Figure 6.10: The throughput of 7-node chain network

four-way handshake procedure. The coding opportunities in such a large network may be less than the coding opportunities in a small network like a 9-node grid network. That is, the EHT problem does not always occur in this case. Although the improvements are not as great as in Fig. 6.11, we still prove our four-way handshake procedure is also suitable for the large networks.

In conclusion, the results in Fig. 6.11, Fig. 6.10 and Fig. 6.12 show the network coding performance gain can be elevated by reducing the collision occurrences. Our design solves the EHT problem in both small and large networks.

Average End-to-end Delay:

We consider the delay issue in this section. The network coding scheme can reduce the average end-to-end delay in a network because packets may be drained out quickly through the encoding procedure. Table 6.1 and Table 6.2 show this condition where the average end-to-end delays of our network coding scheme are

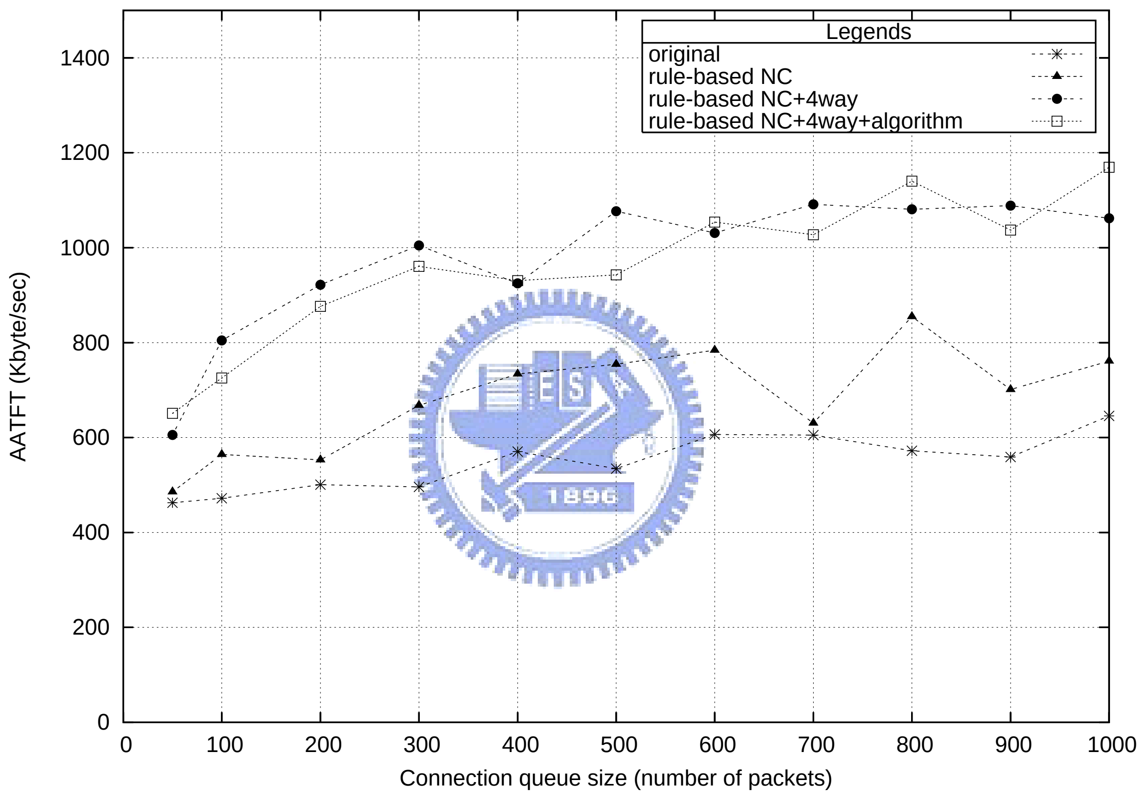

Figure 6.11: The throughput of 9-node grid network

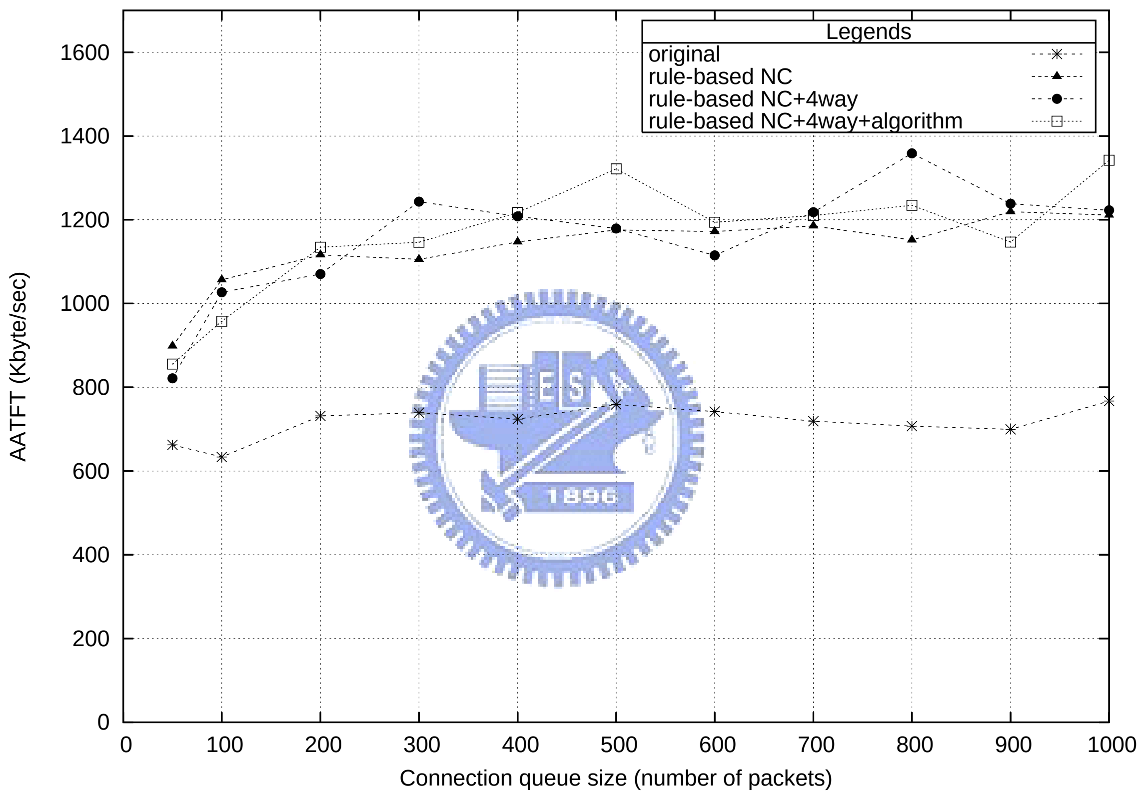

Figure 6.12: The throughput of 25-node grid network

Table 6.4: Average end-to-end delay under different connection queue sizes in 7-node chain networks

connection queue size (packets)	original		rule-based nc		rule-based nc+4way		rule-based nc+4way+algorithm	
	AETED (sec)	Sdv	AETED (sec)	Sdv	AETED (sec)	Sdv	AETED (sec)	Sdv
50	0.317	0.004	0.202	0.015	0.233	0.008	0.223	0.004
100	0.550	0.007	0.374	0.001	0.427	0.004	0.409	0.008
200	1.171	0.091	0.679	0.015	0.784	0.028	0.813	0.004
300	1.908	0.166	1.007	0.036	1.203	0.098	1.057	0.096
400	2.598	0.188	1.309	0.129	1.413	0.098	1.354	0.133
500	3.228	0.249	1.401	0.186	1.916	0.233	1.596	0.001
600	3.867	0.306	1.800	0.019	2.064	0.272	2.217	0.005
700	4.248	0.022	2.019	0.128	2.342	0.125	2.587	0.060
800	4.632	0.092	1.961	0.237	2.508	0.121	2.965	0.092
900	5.031	0.200	2.383	0.314	2.921	0.246	3.300	0.079
1000	5.398	0.303	2.558	0.178	3.150	0.344	3.589	0.406

less than of original WiMAX scheme.

Another delay topic is related to the effects of the four-way handshake procedure and the data schedule holdoff algorithm. The four-way handshake procedure will cost some time to forward the extended confirm messages and lengthen the end-to-end delays. Furthermore, the data schedule holdoff algorithm will postpone the data schedule to prevent collision to occur. This behavior will also increase the end-to-end delays. In Table 6.4, Table 6.5 and Table 6.6, we list the average end-to-end delay and the standard deviation under different connection queue sizes. The results correspond with our expectation explained above, the average end-to-end delay under four-way handshake procedure is larger than under our network coding scheme. Besides, the average end-to-end delay using data schedule holdoff algorithm is larger than only using four-way handshake procedure.

6.5 System Resources Discussion

In the following, we discuss some system issues in the real-world and display the practicability of our network coding scheme.

Table 6.5: Average end-to-end delay under different connection queue sizes in 9-node grid networks

connection queue size (packets)	original		rule-based nc		rule-based nc+4way		rule-based nc+4way+algorithm	
	AETED (sec)	Sdv	AETED (sec)	Sdv	AETED (sec)	Sdv	AETED (sec)	Sdv
50	0.394	0.008	0.388	0.019	0.430	0.023	0.419	0.010
100	0.757	0.074	0.739	0.034	0.828	0.057	0.792	0.019
200	1.450	0.089	1.293	0.181	1.553	0.079	1.379	0.067
300	2.096	0.112	1.698	0.159	2.036	0.069	2.288	0.294
400	2.758	0.311	2.050	0.173	2.556	0.258	2.445	0.216
500	3.406	0.248	3.031	0.149	2.908	0.109	3.086	0.425
600	3.908	0.254	3.564	0.399	4.210	0.326	3.494	0.295
700	4.430	0.305	3.575	0.440	4.319	0.451	4.215	0.253
800	4.908	0.370	3.429	0.586	3.959	0.406	5.898	0.458
900	5.580	0.949	3.402	0.902	4.225	0.456	6.487	0.714
1000	6.088	0.609	4.793	0.809	5.708	0.466	7.073	0.483

Table 6.6: Average end-to-end delay under different connection queue sizes in 25-node grid networks

connection queue size (packets)	original		rule-based nc		rule-based nc+4way		rule-based nc+4way+algorithm	
	AETED (sec)	Sdv	AETED (sec)	Sdv	AETED (sec)	Sdv	AETED (sec)	Sdv
50	1.063	0.233	0.813	0.225	0.903	0.197	0.878	0.188
100	1.965	0.406	1.366	0.272	1.624	0.374	1.584	0.343
200	3.482	0.614	2.458	0.498	2.990	0.602	2.825	0.529
300	4.957	0.971	3.661	0.940	4.199	0.997	4.098	0.787
400	6.199	1.237	4.355	0.899	5.208	1.162	5.373	1.075
500	7.816	1.550	5.581	1.313	6.237	1.287	6.197	1.116
600	9.133	1.804	6.625	1.547	7.797	1.640	7.358	1.541
700	10.894	2.202	7.514	1.520	8.010	1.571	8.165	1.657
800	11.998	2.843	8.268	1.784	9.378	1.679	9.308	2.164
900	12.989	3.360	8.936	2.023	9.857	2.171	9.784	2.519
1000	14.018	3.452	9.040	2.353	10.548	2.650	10.044	1.974

- Memory

  In our proposed network coding scheme, the devices in the network should overhear and store packets which are not sending for themselves. We clean the queuing data periodically to prevent the memory ran out. In addition, there is no need to store a packet which is already used for decoding or is already arrived at the destination. Hence, we will only need to store the packets in flight in a round-trip-time (RTT).

- Power

  Our network coding scheme is based on the IEEE 802.16(d) mesh networks. In this standard, the WiMAX mesh nodes are all fixed and without moving. Therefore, we can assume that the power issue is not a problem where a fixed node should have a stable power supply.

Chapter 7

Future Work

- **The Data Schedule Holdoff Algorithm in Four-way Handshake Procedure**

  Recalled in Section 5.4, in order to insure the extended confirm messages will arrive before the data schedule starting, we look up the next control message transmitting time of all corresponding nodes, and delay the data schedule starting time if it needed. This procedure reduce the collisions successfully as shown in Table 6.3. However, the performance may be reduced by unnecessary protections due to the long back-off of data schedules. If a collision never affects the network coding scheme, it can be ignored without any protection. But it is hard to distinguish a packet whether it needs protection or not.

- **The Election Algorithm when Schedules Overlap in Four-Way Handshake Procedure**

  As mentioned in Section 5.3, the extended confirm handling procedure needs to decide which node can transmit data when a extended confirm schedule is overlapped with existing data schedules. In this thesis, we use a simple method to make the decisions as explained in Section 5.3. However, this method may cause unfair conditions. That is, some nodes always win the election and some nodes are forced to yield the transmission opportunities. The new algorithm must be designed to deal with the fairness issue.

Chapter 8

Conclusion

In this thesis, we propose an easy-to-implemented and easy-to-deployed inter-session network coding scheme that is based on the opportunistic approach. Our proposed scheme need not employ an additional protocol to find coding opportunities or coding structures instead of other researches. Oppositely, our proposed scheme exploits only routing and MAC-layer information to find coding opportunities and coding structures. Using this advantage, it is easier to be realized in a real-life network.

In addition, in this thesis, we point out a new "extended hidden terminal problem," which can frequently occur in a wireless network using network coding. The "extended hidden terminal problem" can result in the failure of network-coded packet decoding, thus significantly decreasing the performance of a network-coded wireless network. We explain why the "extended hidden terminal problem" cannot be solved by the currently-existing wireless network standards, such as the IEEE 802.11(b) and the IEEE 802.16(d) specifications. Moreover, we also propose a bandwidth reservation mechanism that is extended from the IEEE 802.16(d) mesh-mode three-way handshake procedure to prevent the "extended hidden terminal problem" from occurring.

Although our mechanism proposed in this paper is specific to the IEEE 802.16 mesh network, it can be used as an example protocol design to eliminate the "extended hidden terminal problem" for other network-coded wireless networks.

Finally, the simulation results show that the throughput performance of a network using our proposed scheme is better than that of a network using the traditional routing. In addition, the performance of our proposed four-way handshake design is worthy of being further study.

Bibliography

[1] "802.16d 2004. Draft IEEE standard for local and metropolitan area networks - Part 16: Air interface for fixed broadband wireless access systems". May. 2004.

[2] R. Ahlswede, N. Cai, S. R. Li, and R. W. Yeung. "Network Information Flow". *IEEE Transactions on Information Theory*, pages 1204–1216, July. 2000.

[3] P. A. Chou and Y. Wu. "Network Coding for the Internet and Wireless Networks". June. 2007.

[4] P. A. Chou, Y. Wu, and K. Jain. "Practical Network Coding". *51st Allerton Conf. Communication, Control and Computing*, Oct. 2003.

[5] H. Chu. "Simulating WiMAX Mesh Networks over the NCTUns Network Simulator". June. 2006.

[6] A. Eryilmaz and D. Lun. "Control for inter-session network coding". *In Proceedings of the Workshop on Network Coding, Theory and Applications*, 2007.

[7] K.-H. Fang. "A Novel MAC-layer Scheduling Scheme for Improving Qos Support in IEEE 802.16(d) Mesh Networks". June. 2007.

[8] A. A. Hamra, C. Barakat, and T. Turletti. "Network Coding for Wireless Mesh Networks: A Case Study". *In Proceedings of IEEE International Symposium on a World of Wireless, Mobile and Multimedia Networks (WoWMoM'06)*, 2006.

[9] T. Ho, Y. Chang, and K. J. Han. "On constructive network coding for multiple unicasts". *invited paper, 44th Allerton Conference on Communication, Control and Computing*, 2006.

[10] T.-W. Hsu. "Using Steerable Direction Antennas to Improve the Performances of IEEE 802.16(d) Mesh Networks". June. 2007.

[11] S. Katti, S. Gollakota, and D. Katabi. "Embracing Wireless Interference: Analog Network Coding". *In Proceedings of the ACM SIGCOMM*, 2007.

[12] S. Katti, H. Rahul, W. Hu, D. Katabi, and M. Medard. "XORs in the air: practical wireless network coding". In *Proceedings of Sigcomm 2006*, pages 243–254, September. 2006.

[13] A. Khreishah, C.-C. Wang, and N. Shroff. "An Optimization Based Rate Control for Communication Networks with Inter-session Network Coding". *IEEE Infocom, Phoenix, Arizona, USA*, April. 15-17. 2008.

[14] S. Omiwade, R. Zheng, and C. Hua. "Butterflies in the mesh: lightweight localized wireless network coding". *In Proceedings of 4th Workshop on Network Coding, Theory, and Applications (Netcod)*, 2008.

[15] S. Sengupta, S. Rayanchu, and S. Banerjee. "An Analysis of Wireless Network Coding for Unicast Sessions: The Case for Coding-Aware Routing". In *Proceedings of the 26th IEEE International Conference on Computer Communications (INFOCOM 2007)*, pages 1028–1036, May. 2007.

[16] C.-C. Wang and N. Shroff. "On Wireless Network Scheduling with Intersession Network Coding". *In Proceedings of the 42nd Conference on Information Sciences and Systems, Princeton, USA*, Mar. 19-21. 2008.

[17] S. Wang, C. Chou, C. Huang, C. Hwang, Z. Yang, C. Chiou, and C. Lin. "The Design and Implementation of the NCTUns 1.0 Network Simulator". *Computer Networks*, 42(2):175–197, June. 2003.

[18] S. Wang, C. Chou, and C. Lin. "The Design and Implementation of the NCTUns Network Simulation Engine". In *Elsevier Simulation Modelling Practice and Theory*, 2007.

[19] S. Wang, C. Lin, K. Fang, and T. Hsu. "Facilitating the Network Entry and Link Establishment Processes of IEEE 802.16 Mesh Networks". In *Wireless Communication and Networking Conference 2007 (WCNC 2007)*, Mar. 2007.