

## DELAY CONTROL NETWORK CODING TECHNIQUE

Shereen A. M. Ahmed Hamato\*, Sharifah H. S. Ariffin ,  
Norsheila Fisal, Farizah Yunus

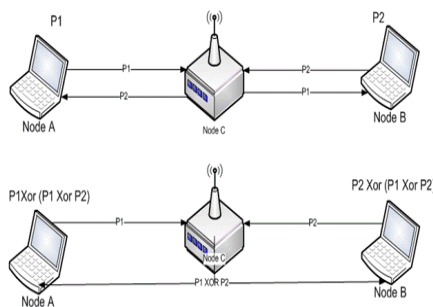
UTM-MIMOS Centre of Excellence in Telecommunication  
Technology, Faculty of Electrical Engineering, Universiti  
Teknologi Malaysia, 81310 UTM Johor Bahru, Johor, Malaysia

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\*Corresponding author  
shereen@fkegraduate.utm.my

### Graphical abstract



### Abstract

Network coding is a technique known to efficiently utilize the bandwidth by exploiting the broadcast nature of the wireless medium. Network coding reduces the number of retransmissions by allowing the relay not only to forward the packets, but to do some logic operation. However, considering the randomness and the asymmetric nature of the traffic in the wireless medium, it is usually very challenging for the relay when receives a packet is whether to hold the packet in order to obtain a network coding opportunity or to rebroadcast the packet directly and eliminate the delay. In this paper, we address this challenge by introducing two schemes; Bandwidth Consideration Scheme (BCS) which considers pure network coding to achieve the maximum improvement in network throughput, and Time Limited Scheme (TLS), which uses the network coding but considers the imposed delay. The results show that, BCS can lead to up to 50% improvement in the bandwidth, however for symmetric flows using pure network coding leads to unbounded delay. On the other hand, TLS noticeably decreases the imposed delay for the symmetric flows and leads to relatively similar improvement in the throughput for asymmetric flows.

Keywords: Network coding, broadcasting, throughput, network congestion

### Abstrak

Pengekodaan rangkaian adalah teknik yang diketahui cepat menggunakan jalur lebar dengan mengeksploitasi penghantaran semulajadi media tanpa wayar. Pengekodaan rangkaian mengurangkan bilangan penghantaran semula dengan membenarkan geganti bukan sahaja untuk menghantar paket, tetapi untuk melakukan operasi logik. Walau bagaimanapun, mempertimbangkan kerawakan dan sifat semulajadi trafik yang tidak simetri dalam medium tanpa wayar, ia biasanya sangat mencabar bagi geganti untuk meramalkan kedatangan paket yang seterusnya, dengan demikian persoalan utama geganti apabila menerima paket ialah sama ada untuk memegang paket bagi mendapatkan peluang pengekodan rangkaian atau penghantaran semula paket secara langsung dan menghapuskan kelewatan. Dalam kertas ini, kita menangani cabaran ini dengan memperkenalkan dua skim; Skim Pertimbangan Jalur Lebar (BCS) yang mempertimbangkan pengekodan rangkaian tulen untuk mencapai kemajuan yang maksimum dalam daya pemprosesan rangkaian, dan Skim Masa yang Terhad (TLS), yang menggunakan pengekodan rangkaian tetapi mempertimbangkan kelewatan yang dikenakan. Keputusan menunjukkan bahawa, BCS boleh membawa kepada peningkatan sehingga 50% dalam jalur lebar, namun bagi aliran simetri menggunakan pengekodan rangkaian tulen membawa kepada kelewatan yang besar. Sebaliknya, TLS nyata mengurangkan kelewatan yang dikenakan bagi aliran simetri dan membawa kepada peningkatan yang agak sama dalam daya pemprosesan untuk aliran tidak simetri.

**Kata kunci:** Pengekodon rangkaian, penghantaran, daya pemrosesan, kesesakan rangkaian

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## 1.0 INTRODUCTION

Network coding is a technique known to efficiently utilize the bandwidth by exploiting the broadcast nature of the wireless medium. Network coding reduces the number of retransmissions by allowing the relay not only to forward the packets, but to do some sort of logic operation on them. A typical network coding operation is shown in Figure 1. Suppose node A needs to send packet  $p_1$  to node B, and node B is sending packet  $p_2$  to node A. Node C in the middle supposed to relay the data of both node A and node B. So in the traditional routing, node C will relay each packet in a separate event. However, using network coding, node C will combine both packets by performing a bitwise XOR operation on them, and sends them in one event, therefore, both of node A and node B will receive the encoded packet. However, both nodes will be able to decode the received encoded packet by XOR-ing it with their own previously sent packet.

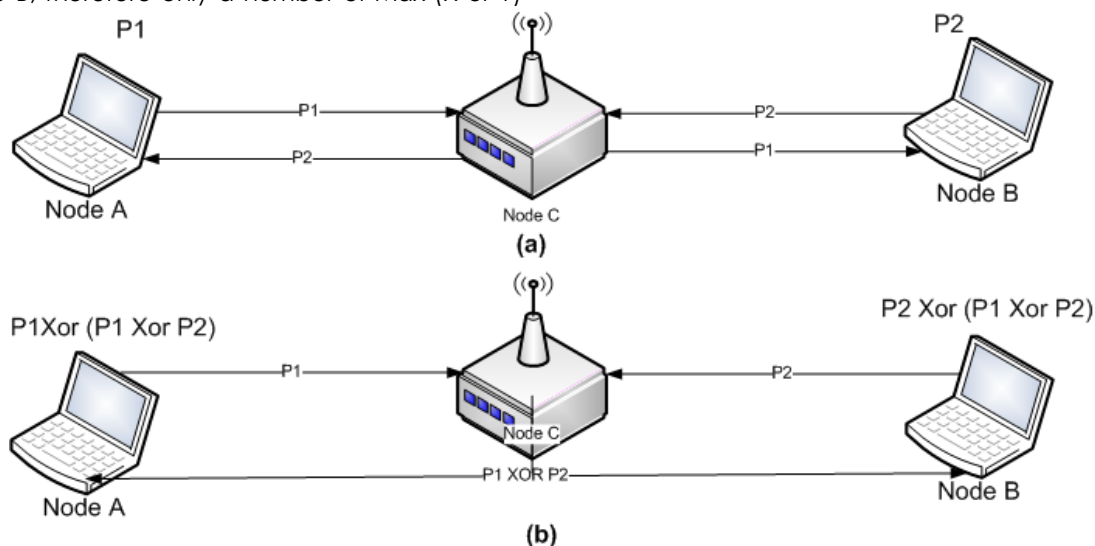
Generally, to more precisely find how much of the bandwidth can be saved in case of the network coding, let us say we have  $X$  packets to be disseminated from node A and  $Y$  packets to be disseminated from node B. In the conventional broadcasting case the relay will need to relay  $X+Y$  packets, if we assume each packet will consume  $1/T$  of the bandwidth, a total of  $(X+Y)/T$  of the bandwidth will be consumed. Assuming the network coding, each packet from node A will be combined with a packet from node B, therefore only a number of  $\text{Max}(X \text{ or } Y)$

(e.g. the higher number of packets) packets will be rebroadcasted. This will consume  $\text{Max}(X \text{ or } Y)/T$  of the bandwidth. Therefore, the improvement in the overall throughput will be  $(\text{Max}(X \text{ or } Y))/(X+Y)$ .

However, if there is the same number of packets from both directions (e.g.  $X=Y$ ), then up to 50% of the consumed bandwidth can be saved when we use the network coding technique.

Nevertheless, this improvement in the bandwidth consumption comes at the cost of the delay. Considering the randomness of the wireless medium, it is usually very challenging for the relay to predict when the next packet is coming, thus the main question for the relay when receives a packet is whether to hold the packet in order to obtain a network coding opportunity or to rebroadcast the packet directly and eliminate the delay.

In this paper will introduce two schemes; Bandwidth Consideration Scheme (BCS) which considers pure network coding, to achieve the maximum improvement in network throughput, and Time Limited Scheme (TLS) that uses the network coding but considers the imposed delay. TLS will obtain all the coding opportunities, however the packet will not wait more than a certain time limit before being released either as a coded packet or as a native (un-coded) packet.



**Figure 1** Typical Network Operation with and without network coding; in (a), without network coding Node C needs to perform two broadcasting events, while in (b) using network coding Node C performs only one broadcasting event, thus the relay performs a less number of broadcasting events when uses the network coding.

## 2.0 LITERATURE REVIEW

Network coding is a technique known to efficiently utilize the bandwidth by exploiting the broadcast nature of the wireless medium. Network coding reduces the number of retransmissions by allowing the relay not only to forward the packets, but to do some sort of logic operation on them.

In literature, there are many applications where network coding has been used. For example:

**Packet Retransmission:** Authors of NCCARQ-MAC[1] propose a protocol coordinates the channel access among a set of relays capable of using network coding. In NCCARQ-MAC upon erroneous packet reception by the destination, the destination will send a retransmission request back to the source, the node that supposed to relay the retransmission request packet to the source, upon the reception of another packet from the source that needs to be relayed to the destination, will encode the retransmission request and the new packet from the source and send them in one event, since the source has a copy of the new packet while the destination has a copy of the retransmission request packet, they will both be able to decode the encoded packet.

**Broadcasting:** Aiming to result in better reliability and loss recovery compare to the traditional protocols [2] and [3], DONC[4] uses the network coding to reduce the twofold effect of the packet loss.

**Multimedia Streaming:** here, the network coding is being used in the peer to peer communication for content distribution purposes [5-9]; it is based on the principle of swarming, wherein the desired file is downloaded in parallel from a number of cooperating peers. The parallel download approach enables the network to achieve improved performance as compared to other peer-to-peer systems in wired network [10].

## 3.0 PROTOCOL METHODOLOGY

Our work assumes a scenario where there are two nodes exchanging information, those two nodes are not in the radio range of each others , and thus the packets need to be relayed at intermediate relay.

In our protocol the relay will maintain a queue to buffer the flow that comes from one source. However, in our proposal, we assume the relay is able not only to forward the message, but actually to perform some kind of logical operations on them. We also assume that the data rates from the two sources are different, that, one source is sending at a rate that is slower than the other. And we assume that the relay is aware of the average rates the sources are sending in.

## 3.1 Relay Behavior:

We propose two schemes for our relay behavior, namely: Bandwidth Consideration Scheme (BCS) and Time Limited Scheme (TLS)

### 3.1.1 Bandwidth Consideration Scheme (BCS):

In this scheme, the relay buffers all the data that comes from the slow source while sending the data that is received from the fast source directly either coded if there is a coding opportunity or un-coded if there is no chance to code.

The protocol work as follows: when the relay receives a packet, it will check to see whether the packet is from the slow rate source or from the fast rate source, if the packet is from the slow rate source, the packet will be buffered into the queue, otherwise , if the packet is from the fast rate source, the relay will first check the queue for a coding opportunity, if the queue is non-empty , the header packet of the queue will be encoded with the new arrived packet and sent out as one packet. Otherwise, if the queue is empty, then the packet will be sent directly without any delay. Figure 2 shows the steps to be followed by this scheme.

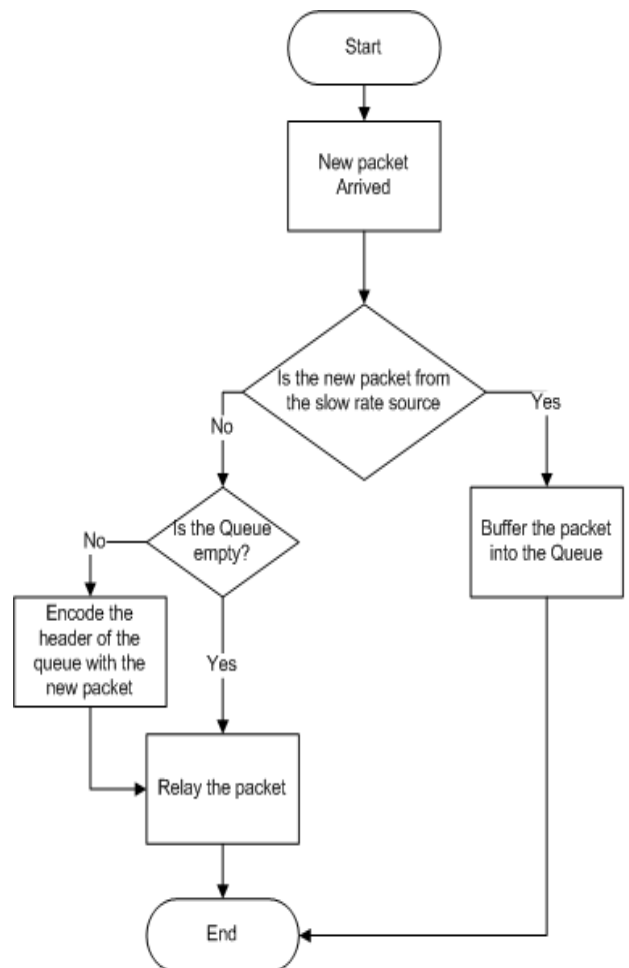


Figure 2 Flowchart of BCS.

**3.1.2 Time Limited Scheme (TLS):**

In order to avoid the long delay for the packets that belong to the slow rate source, this scheme introduces a delay limit to each packet that enters the queue. Here the protocol will again follow the same steps in DCS scheme, except, that for each packet that enters the queue, the relay will set a timer for  $T_{max}$  time units, after this time, if the packet is still in the queue the packet will leave the queue and get relayed directly without any further delay. Figure 3 shows the step to be followed by such scheme.

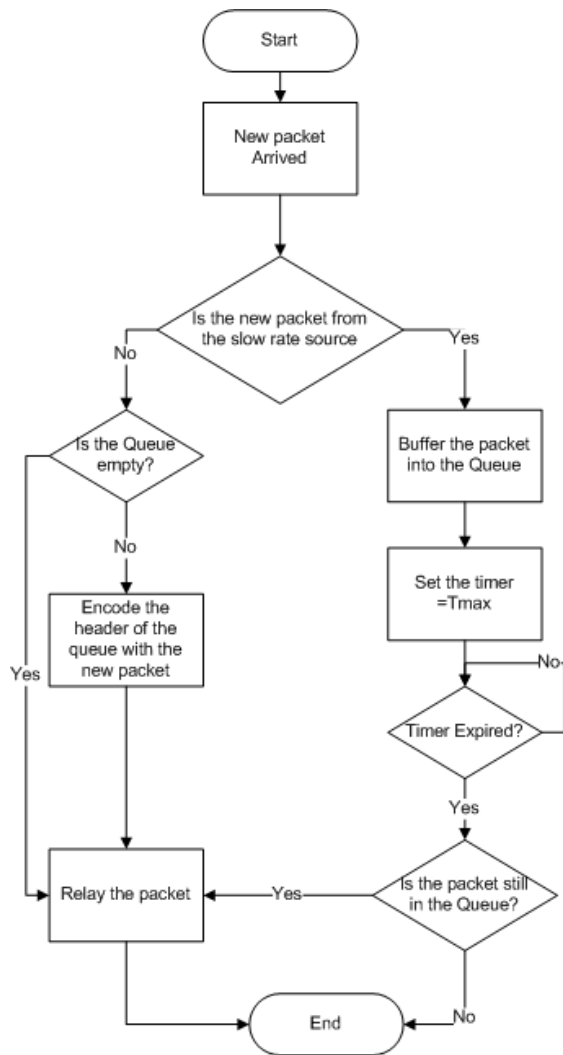


Figure 3 Flowchart of TLS.

**4.0 SIMULATION SETUP**

To simulate scenarios such as the ones are described in section 3 and by Figure 2 and Figure 3. We implemented the protocol on a network simulator (i.e. NS3 version 19), and experimented using IEEE

802.11. The data rates for source1 and source2 was generated randomly (following Poisson distribution) with averages  $\lambda_1$  and  $\lambda_2$  respectively, however,  $\lambda_2$  is assumed to be the slow rate source and set to be 0.1 while  $\lambda_1$  has been changed throughout the reading from 0.1 to 0.6 to obtain different results. To observe to the effect of  $T_{max}$  on the improvement in the throughput as well as the delay, two values of  $T_{max}$  in the TLC scheme have been considered. The simulation continued for up to 2000 seconds (simulation time) before the results recorded. Table 1 shows the simulation parameters setting for our protocol.

Table 1 The simulation parameters

Simulation Parameter	Value		
Radio Range	300 meters		
MAC Protocol	IEEE 802.11p (WAVE)		
Packet size	1000 bytes		
Physical layer setting	Modulation type	Data Rate	Channel BW
	OFDM	6 Mbps	10 MHz

**Performance Metrics:**

Our protocols are designed to achieve a better bandwidth utilization while control the imposed delay. Hence, numbers of performance metrics have been chosen to reflect the performance of our protocols as follows:

**Hop delay:**

This is the average time the packet from the slow source might spend at the hop.

**Percentage of Improvement in the Throughput:**

By this we mean how much of the bandwidth has been saved, and we calculate it as follow:

$$\frac{\text{(Total number of packets scheduled for rebroadcasting)}}{\text{(Actual number of rebroadcasted packets)}}$$

**5.0 THE RESULTS AND DISCUSSION**

**Hop delay:**

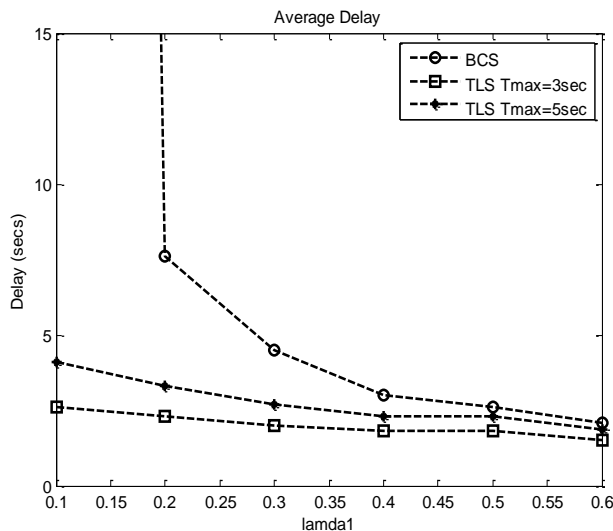
Figures 4 shows the imposed delay in BCS scheme and two cases of TLS scheme (e.g.  $T_{max}=5$ ,  $T_{max}=3$ ). BCS scheme imposed unbounded delay in case of symmetric flows (e.g.  $\lambda_1=0.1$ ), hence, pure network coding is not recommended for symmetric flows. However, for BCS scheme, the delay decreases as the ratio between the two rates increases ( $\lambda_1/\lambda_2$ ), TLS scheme greatly bounded the delay in the symmetric flows case. Moreover, the delay stays relatively stable for all the values of  $\lambda_1$ . As expected in TLS scheme, the lower the  $T_{max}$  the lower the hop delay, for example, for  $\lambda_1=0.1$ , when  $T_{max}=5$  and

$T_{max} = 3$  the recorded delay was 4.1 and 2.6 respectively.

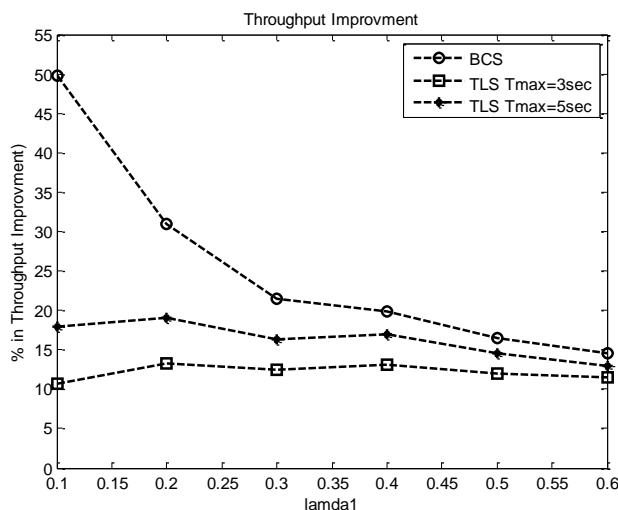
**Percentage of Improvement in the Throughput:**

The throughput improvement hits its upper bound in case of the symmetric rates for the BCS scheme (e.g. 50%). However, it should be kept in mind that, this leads to unbounded delay as shown in Figure 4. In the case of asymmetric flows, for BCS, the throughput improvement decreases as the ratio between the two flows increases.

Although, TLS scheme recorded lower throughput improvement in the case of symmetric flows, yet this scheme records a bounded delay. However, by reading in both Figure 4 and Figure 5, we see that as we increase  $T_{max}$  the throughput will increase, however, the hop delay will also increase, and thus  $T_{max}$  should carefully be chosen to meet the demand of certain QoS.



**Figure 1** The imposed delay in secs Vs lambda1 for BCS and TLS  $T_{max} = 3$  and  $T_{max} = 5$ ,  $\lambda_2 = 0.1$



**Figure 2** The percentage in the throughput improvement vs  $\lambda_1$ , for BCS and TLS  $T_{max} = 3$  and  $T_{max} = 5$ ,  $\lambda_2 = 0.1$

**6.0 CONCLUSION**

Network coding is a promising technique to enhance the throughput of the network, however the main challenge will be for the relay to decide whether to hold the packet to obtain coding opportunity or send it directly and improve the throughput, in this work we introduce two schemes to achieve better performance in term of the delay and the throughput improvement. The result shows that, in symmetric rate flows, pure network coding leads to unbounded delay and thus it is un-recommended. For symmetric flows In TLS scheme, we control the imposed delay by setting an upper bound to the hop delay. In TLS scheme the delay is controlled and an improvement in the throughput has been recorded. The results also show that, as the difference between the data rates of the sources increases the recorded delay decreases. In our future work we are considering the network coding in scenarios where the nodes are mobiles, which will impose more challenges to the current scenario.

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**References**

- [1] Antonopoulos, A., C. Skianis, and C. Verikoukis. 2013. Network Coding-Based Cooperative ARQ Scheme For VANETs. *Journal of Network and Computer Applications*. 36(3):1001-1007.
- [2] Ahmed, S.A.M., S.H.S. Ariffin, and N. Faisal. 2014. Survey on Broadcasting in VANET. *Research Journal of Applied Sciences, Engineering and Technology*. 7(18).
- [3] Hamato, S., Ariffin, S., Faisal, N. 2014. Contention Free Time Efficient Broadcasting Protocol for Safety Applications in VANETs. *International Review on Computers and Software (IRECOS)*. 9(11): 1923-1931.
- [4] Mirani, F.H.; Busson, A.; Adjih, C. 2013. "DONC: Delay-based Opportunistic Network Coding Protocol," *Ad Hoc Networking Workshop (MED-HOC-NET), 12th Annual Mediterranean* : 34-41. 24-26 June 2013 doi: 10.1109/MedHocNet.2013.6767407.
- [5] Park, J.-S., et al. 2006. Emergency Related Video Streaming In VANET Using Network Coding. *In Proceedings Of The 3rd International Workshop On Vehicular Ad Hoc Networks*. ACM: Los Angeles, CA, USA. 102-103.
- [6] Ahmed, S. and S. S. Kanhere. 2006. VANETCODE: Network Coding To Enhance Cooperative Downloading In Vehicular Ad-Hoc Networks, *In Proceedings Of The 2006 International Conference On Wireless Communications And Mobile Computing*. ACM: Vancouver, British Columbia, Canada. 527-532.

- [7] Lee, U., et al. 2006. Code Torrent: Content Distribution Using Network Coding In VANET, *In Proceedings Of The 1st International Workshop On Decentralized Resource Sharing In Mobile Computing And Networking*. ACM: Los Angeles, California : 1-5.
- [8] Li, M., Z. Yang, and W. Lou. 2011. CodeOn: Cooperative Popular Content Distribution for Vehicular Networks using Symbol Level Network Coding. *IEEE J.Sel. A. Commun.* 29(1): 223-235.
- [9] Yang, Z., M. Li, and W. Lou. 2010. CodePlay: Live Multimedia Streaming In Vanets Using Symbol-Level Network Coding, *In Proceedings of the The 18th IEEE International Conference on Network Protocols, IEEE Computer Society*. 223-232.
- [10] Shabbir Ahmed and Salil S. Kanhere. 2006. VANETCODE: Network Coding To Enhance Cooperative Downloading In Vehicular Ad-Hoc Networks. *In Proceedings Of The 2006 International Conference On Wireless Communications And Mobile Computing (IWCMC '06)*. ACM, New York, NY, USA : 527-532. DOI=10.1145/1143549.1143654 <http://doi.acm.org/10.1145/1143549.1143654>.