IEEE 802.16 시스템에서 양방향 트래픽을 위한 네트워크 코딩 기법

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Network Coding for Bidirectional Traffic in IEEE 802.16 Systems

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요 약

본 논문에서는 네트워크 코딩 기법이 IEEE 802.16 계열의 무선 통신 시스템에 어떻게 적용되는지에 대해 다룬다. 또한, 본 논문에서는 네트워크 코딩의 overhearing 문제를 피하기 위해, 통신하고자 하는 양단의 단말들이 중계기를 통해 정보를 서로 교환하는 양방향 트래픽에 주안점을 두고 있다. 이러한 양방향 트래픽은 TCP 프로토콜이 혼잡 제어와 ACK을 기반으로 하는 에러 복구를 수행하는 인터넷 환경에서 주로 관찰된다. 그러므로 네트워크 코딩 기법의 적용을 통해 무선 인터넷의 스펙트럼 효율을 향상시킬 수 있을 것이라 예상되는 바이다. 본 논문에서의 모의실험 결과는 네트워크 코딩 기법을 적용하였을 때, 기존의 무선 중계 시스템과 비교하여 평균적으로 36 퍼센트의 수율 향상을 가져올 수 있음을 보여준다.

Key Words : Network Duality, Power Control, Optimization, Interference Channel, Lagrangian

ABSTRACT

In this paper, we investigate how the IEEE 802.16 based wireless system can adopt the network coding. To avoid the problem of overhearing, we focus on the bidirectional traffic, where each end node exchanges network coded data over a relay node. The bidirectional traffic is usually observed in Internet, where TCP makes congestion control and error recovery based on the acknowledgement from the opposite direction. Thus, enhancing the spectral efficiency of wireless Internet through the network coding is expected. Our simulation with realistic radio characteristics and TCP-like traffic shows that the network coding improves the throughput by an average of 36 percent compared to the simple relay case.

Ⅰ. Introduction

Network coding (NC)[1], a fine method to squeeze the link resource has attracted interest of researchers and industries. Although the example that shows the (network) coding gain starts from the noiseless wired multicast network, applications in wireless networks get more important nowadays and are

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investigated (see [2] and literature therein). It is theoretically shown in [3] that the linear code can achieve the maximum throughput bound of the multicast network. Accordingly, most of the research in this area uses a simple linear code, e.g., XOR operation, for encoding/mixing independent streams into one. Although there are physical layer network coding schemes[4], most operation of NC is done in upper layers; for example in [2], the authors put NC between media access control (MAC) and network layers, which differentiates it from the physical layer coding, and makes it called as network coding.

Despite many known superiorities of NC, however, there is also practical requirement that the receiving node should have sufficient information to decode the (network) coded packet. For example, when a packet is encoded by XOR operation with n independent packets, the receiver should a priori know the n-1 packets except the one that it has to receive. Different from the original multicast case of wired networks, in the case of wireless networks, this requirement should rely on broadcasting nature of the wireless channel. That is, the receiver node may overhear the packets transmitted by other nodes. However, overhearing might not be perfect and consequently, there is less NC gain in sparse networks, where the nodes are located with long distance away. The network coded packets could not be decoded at each intermediate nodes owing to the deficiency of packets.

As the coded packet is targeted to multiple receiving nodes at every hop, the number of receiver nodes at a given instant increases when compared to the non-coded network. This would become problematic especially in wireless environments, because the average number of receivers that should be taken care of increases. This often triggers retransmission of coded packets, which may reduce the overall performance of the network[5].

The above prerequisite, overhearing, could be ignored in bidirectional traffic embedded in the linear topology, in which two end nodes exchange respective data through intermediate relay nodes. In this traffic, two original packets from opposite directions are mixed into one. One of both packets necessary for decoding the mixed packet at relay nodes is already known; these known packets are what relay nodes transmitted earlier. As a simple example, we can think of the Alice - access point - Bob topology depicted in [2].

The main purpose of this paper is twofold: First, we focus on the application of NC to the bidirectional traffic. We deal with a network supported by TCP (Transmission Control Protocol), the transport layer of Internet. Secondly, we focus on a practical system to see how it can be modified to support NC. For this, we investigate how IEEE 802.16j network, an OFDMA-based multihop relay system[6], can be modified to support NC.

With the backward compatibility to the other IEEE 802.16 OFDMA based variations (e.g., commercially known as WiMAX), IEEE 802.16j is designed to basically support multi-hop relay.1) In IEEE 802.16j, relay stations (RS) are introduced in order to enhance the cell coverage and system capacity of the networks. In IEEE 802.16j specifications, there are two scenarios with respect to the introduction of RS: one is called the transparent mode, where RS amplifies only data channels to help a mobile station (MS) within coverage of the base station (BS) receiving data reliably (throughput improvement).

The other is called the non-transparent mode, where RS relays both control and data channels to MS out of the BS coverage (coverage enhancement). In the non-transparent mode of two-hop relay, RS distributively controls RS-MS links on behalf of BS, unlike the transparent mode, in which BS controls the RS-MS link. In this paper, we focus on the non-transparent mode of two-hop relay since the introduction of RS mainly aims at coverage enhancement of BS.

NC can be adopted by both RS and BS in IEEE 802.16j. In NC-RS case, RS can code (mix) the up- and downlink data into one and can multicast the

1) Although a network coding scheme has been proposed to the standardization, it was not adopted in IEEE 802.16m to our best knowledge. However, since IEEE 802.16m standardization in [7] also supports relay mode, our proposed NC scheme can be adapted to IEEE 802.16m.

coded data toward MS and BS. In NC-BS case, two MSs can communicate with each other via BS. Similarly, BS can code the bidirectional traffic from both MSs, as if it is RS in the NC-RS case. As a result, we can save the network resource (e.g., time or frequency, etc.) by reducing the number of up- and downlink transmission of RS and BS.

There have been researches on throughput enhancement by adopting NC into the cellular system. However, they treat NC adopted in either downlink[8] or uplink[9]. They do not consider bidirectional traffics of uplink and downlink, simultaneously. Rather, it is assumed that there are multiple recipients of the same signals in either uplink or downlink phase. Different from these works, NC application for the bidirectional up- and down traffic in cellular systems is explored in [10]. However, investigation in [10] is focused on RS selection and power management, rather than practical aspects that we are considering in this paper.

The rest of the paper is organized as follows: In the next section, we discuss how the TCP-like bidirectional traffic can benefit from NC. In Section Ⅲ, we propose our modified frame structure of IEEE 802.16j so that it can adopt NC to support the bidirectional traffic. Section Ⅳ contains our simulation results on NC over TCP-like traffics. Finally, Section Ⅴ concludes the paper.

Ⅱ. TCP, Bidirectional Traffic and Network Coding

Network coding, when applied to the bidirectional traffic case (i.e., two end nodes exchange data traffic in a linear topology), can avoid the problem of overhearing. A question is if there are ample examples of such a bidirectional case. For this, we can think of applications supported by TCP. TCP is connection-oriented and uses feedback packets, known as ACK (acknowledgement), to make error recovery. This means that for a data packet for one direction, there corresponds an ACK packet from the opposite direction. In a TCP packet, data is usually piggy-backed with ACK information. In this sense, we can see that TCP is a transport protocol that has bidirectional traffic between both end nodes. Considering the popularity of TCP, we can imagine applicability of NC to many Internet services.

On the other hand, TCP has been an issue in wireless environments. The main problem in the wireless TCP is how to enhance the end-to-end throughput, which, unfortunately, still suffers from low spectral efficiency (bits/hertz). TCP was originally designed for wired Internet, and may need modification for using it for wireless links.

The fundamental conflict between TCP and wireless links comes from the congestion control of TCP. Compared to the wireless links, Internet services are based on relatively reliable wired links. Within TCP, error recovery between two communicating end nodes is done by retransmission of the same packet previously detected as being erroneous. The retransmission at the source node is based on the ACK generated by the receiving node.

In the wired network, most of the packets are delivered correctly to the receiving node, but the delay of the packet varies depending on the congestion level of the network. This means that most of the packet error is due to the packet delay, and is caused by congestion level of the network. Thus, TCP couples the error recovery and the congestion control into one; when there occurs an error in a packet (mostly packet delay) at the receiving node, TCP reduces the window size that determines the number of consecutive packets that the source node can send, without getting ACKs from the receiving node. Variations of TCP such as Reno, Tahoe and Vegas are mostly from the aspect of the congestion control.

On the other hand, in the wireless environments, the quality of link fluctuates rapidly, known as wireless fading phenomenon. As the fading becomes severe, the packet error will increase. That is, in the wireless TCP, the packet error is resulted from two distinct reasons: fading and congestion. Thus, the classical congestion control within TCP may under-utilize the bandwidth, by accepting the instantaneous radio fading as symptom of traffic congestion.

Over the decade, there have been many ideas proposed to distinguish the error caused by wireless fading from the one by network congestion, of which main motivation is to enhance the performance of the wireless TCP. However, we did not observe any significant breakthrough with respect to this issue yet.

In TCP, when ACK is not received by the source node until some predefined time (timeout), the source node starts to retransmit the corresponding packet, followed by the other consecutive packets that may have already been received correctly. Even if the corresponding packet was received correctly, the ACK packet may be delivered lately or erroneously to the source node, due to the vulnerability of wireless links of the opposite direction. In this particular case, there might be poor utilization of the bandwidth, by retransmission of the same packets that have already been successfully received and by reducing the congestion window size.

In [11], the authors proposed a new acknowledgement mechanism that plays a key role in incorporating NC into the congestion control for TCP. Furthermore, the authors in [12] proposed a revised protocol that resolves the asynchronism problem between data transmission and decoding operation in [11].

However, [11] and [12] requires changes to the TCP protocol. To overcome this, the authors in [13] proposed a network coding scheme without the need to modify the TCP layer. In [11-13], benefits of using the network coding in TCP are in quick delivery of ACK packets to the source node, by encoding the packets of one direction and ACKs for the opposite direction.

In this paper, we will explore the performance gain from using the network coding in TCP, in particular when the location of relay node is asymmetric like [14].

Ⅲ. Network Coded Transmission in IEEE 802.16 Networks

We start with IEEE 802.16j specifications (the latest document)[6]. Compared to the conventional systems such as IS-95, 3GPP (LTE) and WiMAX etc., introduction of relay stations brings merits such as to lengthen the cell coverage and to strengthen the throughput and system capacity. These profits are originated only from the relay protocol itself. However, NC which is available in relay-enabled communication systems brings more benefits than what just the relay protocol does. Let us look into IEEE 802.16j specification, a multi-hop relay system.

There are two available modes for communication between MS and BS. The first is the direct transmission mode (A1~A2) and the other is relay transmission mode (B1~B4) as shown in Fig. 1. In our scenario, MSs using relay transmission are located outside the coverage of BS, where RS relays both of control and data channels to MS.

The non-transparent frame structure of IEEE 802.16j is described in Fig. 2. (a)[6]. In this frame structure, every up- and downlink traffic from RS should use their own resources exclusively. In other words, the uplink traffic from RS to BS2) occupies the resource (symbol time and channel), which could not be assigned to the downlink traffic from RS to MS.

To adopt NC at RS between MS and BS, we

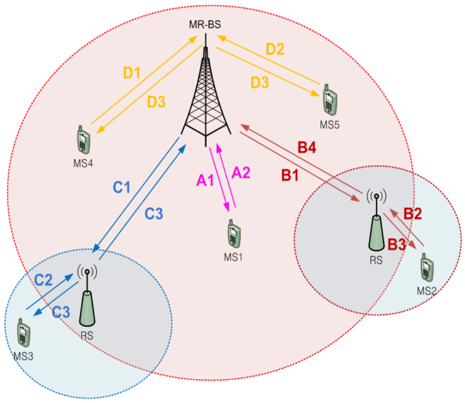

Fig. 1. Scenarios for the network coded transmission mode. In this example, A, B, C and D denote direct transmission mode, relay transmission mode, network coded transmission mode (1) and network coded transmission mode (2), respectively.

2) In IEEE 802.16j specifications, MR-BS denotes the base station that supports a multi-hop relay. For a simplicity, however, we use the term of BS instead of MR-BS.

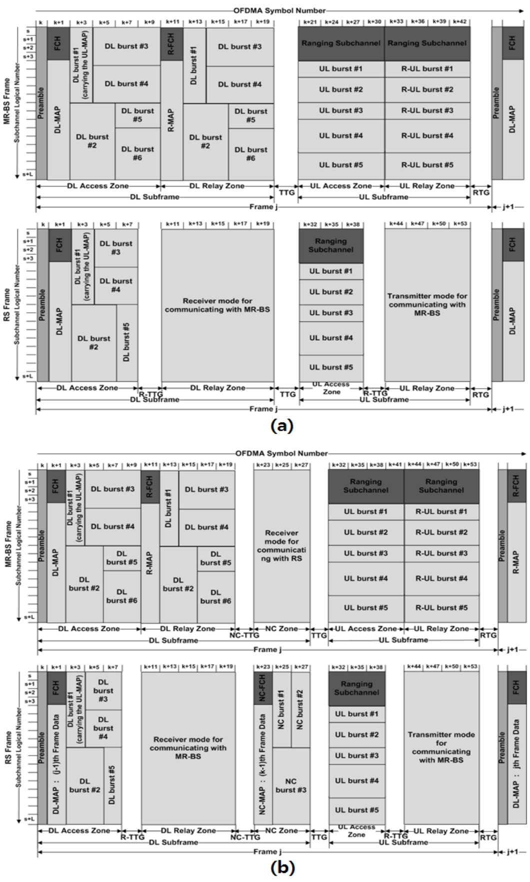

Fig. 2. (a) Frame structure in non-transparent IEEE 802.16j specification [6] (b) Proposed frame structure combined with network coding.

propose that a NC zone is additionally included in the downlink subframe as shown in Fig. 2. (b). In the uplink subframe, MS and RS operate in the transmitting mode and BS operates in the receiving mode. The reason why NC zone is in the downlink subframe is as follows: If NC zone is placed on the uplink subframe, MS has to operate in the receiving mode on its uplink phase. This is against the backward compatibility of MS to IEEE 802.16d/e (WiMAX) specifications. Moreover, there exists an additional transmit/receive transition gap (TTG), i.e., NC-TTG, for the frames of BS and RS due to the NC zone.

Introducing NC zone as in the proposed frame structure, we have to define NC-MAP[3] that includes the information on resources to be assigned for the coded data. Due to the broadcasting nature, the same subchannel and symbol time should be assigned to MS and BS. There could be a question on the efficiency of using additional MAP (i.e., NC-MAP), instead of using DL-MAP of RS for including NC resource information. Especially, every MAP such as DL-MAP and R-MAP is usually modulated in a low-order for reliable transmission. In spite of this probable inefficiency, NC-MAP should be separated from DL-MAP, because, when DL-MAP of RS is transmitted, BS is also in the transmission mode for its DL-MAP. For receiving network coding information, BS should be also in the receiving mode. For this reason, we have a separate NC-MAP in RS.

NC-MAP can be composed by RS or BS. We assume that RS composes NC-MAP so that RS controls the network coding, independent of BS. For this reason, we call this way of composing NC-MAP as distributive scheduling. RS transmitsan NC-burst to both BS and MS, simultaneously. NC-MAP plays a role of DL-MAP for MS and R-MAP for BS, respectively. For the backward compatibility to IEEE 802.16, we propose to utilize the original DL-MAP data structure of RS for NC-MAP, by modifying DL-MAP IEs (Information Element).

CID (Connection Identifier) in DL-MAP IE represents the transmission method such as broadcast, multicast, or unicast. RS sets this CID values in NC-MAP IEs to the multicast transmission mode and adds the address of BS to the set of multicast addresses.

Ⅳ. Simulation Results

To get insights on performance of network coding, we compare the proposed scheme with a two-hop relay OFDMA system. Simulation parameters from [15] and [16] are summarized in Table 1.

We made a system level MATLAB-based simulator using EESM (exponential effective SINR

3) Generally, MAP consists of information about resource (symbol time and channel) allocation for UL/DL transmissions.

Table 1. System parameters

Parameter	Value
system bandwidth	10 MHz
FFT size	1024
OFDM symbol duration	102.9 μs
duplex mode	TDD
channel model between BS and RS	Ped A, 3 Km
channel model between RS and MS	Veh A, 60 Km
path loss model	$128.1+37.6\log_{10}R$, R in Km
BS power	43 dBm
RS power	43 dBm
MS power	23 dBm
thermal noise power	-174 dBm
channel coding	CTC
packet size	24576 bits
ACK timeout	10 ms

mapping) method[17], which maps an instantaneous SINR value in OFDM channel (γ_k) to a SINR value in AWGN channel (γ_{eff}) by the following equation:

$$\gamma_{eff} = -\beta \ln\left(\frac{1}{N}\sum_{k=1}^{N}\frac{\gamma_k}{\beta}\right), \qquad (1)$$

where β is an experiential factor depending on the modulation and coding scheme (MCS) level, and N denotes the number of subcarriers. All MCS levels and the parameter β for each MCS level utilized in our simulations are taken from [18]. The throughput of any given OFDM channel can be estimated by γ_{eff} and BLER (Block error rate) tables for all MCS levels in AWGN channel.

The topology of our simulations is the one, where BS, RS and MS are laid in line. The distance between BS and RS is 700 meters and that between MS and RS is 150 meters.

Figs. 3 and 4 illustrate data flows for the network coding and the simple relay frame structure, respectively. For simplicity of the simulation, we made assumptions on the retransmission as follows: When RTT (round trip time) of each packet is longer than the fixed ACK timeout, retransmission

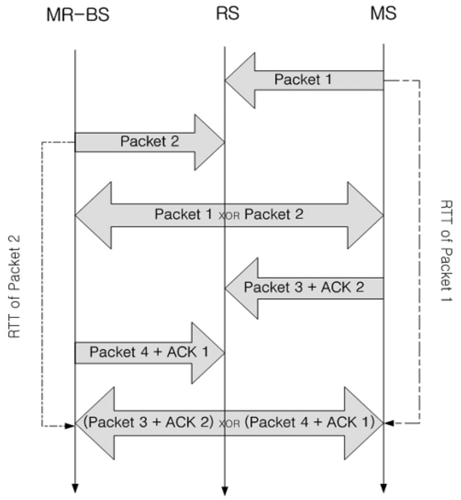

Fig. 3. The data flow of network coded frame structure.

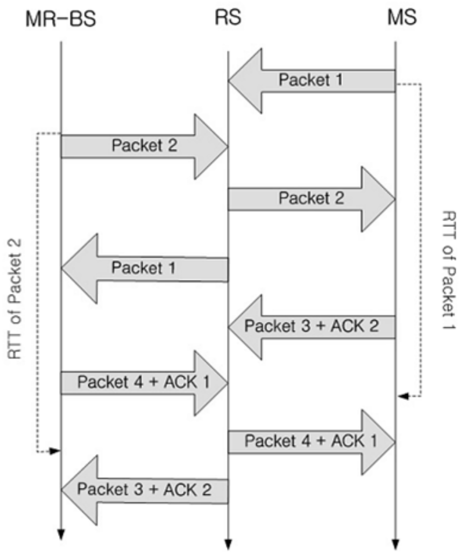

Fig. 4. The data flow of simple two-hop relay frame structure.

of the same packet happens immediately. We assume that the packet size of DL and UL traffics is fixed and one unit frame is finished when DL and UL packets are transmitted completely to each destination.

Table 2 contains the results of each scenario mentioned above. The RTT timeout rate depicts the in-time rate of ACK, i.e., denoting, on average, the how many times the RTT is within the fixed

Table 2. Simulation results

system	RTT timeout rate	throughput
network coding (DL)	0.986	5.22 Mbps
network coding (UL)	0.781	4.13 Mbps
simple relay (DL)	0.694	3.12 Mbps
simple relay (UL)	0.813	3.66 Mbps

timeout. From Table 2, we can predict that the RTT of downlink traffic is shorter than the uplink case in the network coding scenario. Therefore, the downlink rate is larger than the uplink in the network coding scenario.

In the simple two-hop relay scenario, on the other hand, the uplink rate is larger than the downlink case. This is because the transmission time between BS and RS (700 meters) is longer than that between MS and RS (150 meters).

In Fig. 4, we count that the number of BS-RS transmissions is four in the downlink RTT, whereas it is three in the uplink.

Unlike the downlink case, the RTT in-time rate of the network coded uplink is lower than the simple relay uplink scenario. In Figs. 3 and 4, the gain of the network coded uplink (comparedto simple relay) comes from difference of the first frame duration (the number of transmissions is 4 → 3). In the second frame, however, the simple relay needs 3 transmissions and the transmission time from RS to MS is shorter than that of the network coding.

This is because, in the network coded transmission of Fig. 3, RS adjusts its data rate to BS-RS, rather than MS-RS, to guarantee the QoS at the worst link. The relay should ensure a reliable multicast of the network-coded packet to all destinations[19]. The data rate of the relay should be selected according to the destination with the worst channel condition to ensure the reliable multicast.

In the wireless multicast, each channel condition between a relay and multiple destinations is different due to the characteristics of wireless medium, i.e., path-loss, shadowing and multi-path propagation. Therefore, there might be less network coding gain in wireless networks, where each channel between the relay and multiple destinations, i.e., relay channel, is asymmetric, which is the case of our simulation setting.

Table 2 also represents the throughput of each scenario. The throughput is calculated by excluding the retransmission due to ACK timeout. As expected, the network coded downlink throughput has superior performance to the simple relay scenario. The interesting fact is that the network coded uplink throughput is higher than that of the simple relay scenario, even though the RTT timeout rate is lower. This is owing to the shorter frame duration of the network coding than the simple relay; on average, simple relay (5.48 ms) and network coding (4.497 ms) from our simulations. Overall, we see that the network coding improves the up- and downlink throughput by an average of 36 percent compared to the simple relay case. This improvement completely compensates the overhead to have the network coding zone in IEEE 802.16 frame structure in Fig. 2. (b). Note that the improvement would be even more significant when the relay is positioned in the middle between BS and MS, so that the relay channels are rather symmetric.

Ⅴ. Conclusions

In this paper, we investigated how the OFDMA based relay system, e.g., IEEE 802.16j can support the bidirectional traffic between BS, RS and MS, with the network coding enabled at RS. The bidirectional traffic is usually observed in Internet, where TCP makes congestion control and error recovery based on the acknowledgement from the opposite direction. For the purpose, we suggested a new frame structure and signal flows that have backward compatibility to the other IEEE 802.16 systems. When the location of relay node is asymmetric, our simulation with realistic radio characteristics and TCP-like traffic was very encouraging, in that the network coding will be beneficial in practical systems. Throughout the paper, we considered the two-hop relay which is practically attractive, but the generalization of the proposed scheme to the multihop relay would be possible.

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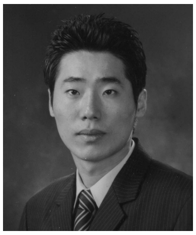

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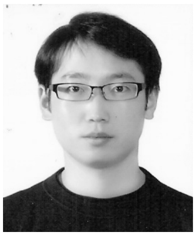

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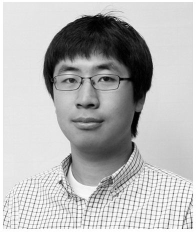

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<관심분야> Radio resource management, random media access control, cognitive radio system

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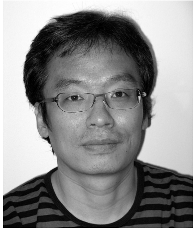

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