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Impact of Channel Bonding on Network Performance in Real-Time Applications

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Abstract*: *This paper investigates the impact of channel bonding property provided in wireless technology on performance in real-time applications. IEEE 802.11n is an amendment to the IEEE 802.11 Wireless Local Area Network (WLAN) standard, which aims to extensively improve network throughput over legacy WLANs. This new network technology provides a better performance for general Internet applications such as web service and file transfer. However, the recent network measurements show that real-time application traffic is consistently increasing in the Internet. Real-time applications such as Voice over IP (VoIP) or video conferencing requires distinct performance metrics compared to the general Internet services in that they prioritize delay, latency, and delay jitter rather than network throughput. This paper investigates how such real-time applications perform in IEEE 802.11n WLANs. Our indoor experiments show that 802.11n basically supports better service than the previous WLAN standards. The channel bonding technique in 802.11n further improves the performance even under mobile conditions.

Keywords*: *Real time transmission, channel bonding, IEEE 802.11n, sensor network.

I. INTRODUCTION

IEEE has approved the 802.11n high data rate wireless LAN (WLAN) standard [1]. This new technology provides enhanced network throughput over previous standards, such as 802.11a/b/g, with a significant increase in the maximum raw OSI physical layer (PHY) data rate from 54 Mbps to a maximum of 600 Mbps. The maximum throughput is achieved with the maximum of four spatial streams using a 40 MHz-wide channel. In practice, however, the maximum data rate of 600 Mbps may not be achievable due to dynamic wireless channel status. It is expected that 200-300 Mbps of average data rate is a realistic number.

As the wireless technology gets popular, more end-point connections are being replaced with WLAN. It is not unusual anymore to connect the Internet via WLAN at home, school, office, and even on a street. WLAN has advantages of easy deployment, scalability, and mobility compared with the conventional wired LAN (e.g. Ethernet). WLAN also eliminates installation of complex and expensive cables, which gets rid of limitation of the cable length. Connections can be maintained while nodes are moving within the communication range. Although a user moves out of the communication range, Internet access is provided seamlessly if multiple APs support Handover service. Since the three advantages are more beneficial than higher data rate from users' perspective, wireless is getting more preferred today although it provides only half data rate compared to the wired. The maximum data rate of WLAN was 54 Mbps until 802.11n technology is developed, and the current most widely deployed Ethernet standard provides at most 100 Mbps data rate. Now, as the 802.11n provides higher data rate than 100 Mbps, it is expected that WLAN can replace most general wired LAN environments except some areas where high reliability and security are required. At this point, we may have a question. Can 802.11n really replace current Ethernet? It is true that 11n technology provides higher data rate than the conventional Ethernet. However, the answer cannot be "yes" because different Internet services have different traffic characteristics. Throughput is not the only criterion for performance evaluation. In real world, real-time application traffics, such as VoIP, video conference, and online game, are becoming dominant. Those traffics have different patterns from general file transfer in that they prioritize delay, latency, and delay jitter rather than network throughput. The above applications also have almost symmetric bi-directional data flows, while down-link traffic is much larger than up-link traffic in conventional Internet applications. The primary goal of newly adopted techniques (i.e. MIMO, Channel Bonding, Frame Aggregation, and etc.) is to improve data rate. Previous researches show that 802.11n provides more network throughput than 802.11a/b/g in various environments [2-5]. However, performance measurements with real-time applications which represent more realistic Internet traffic have not been done extensively. In previous WLAN technologies such as 802.11b/g, real-time applications are not supported well as users increase due to frequent packet generation which leads severe contention in wireless link.

Performance of 802.11 has been investigated based on experiment and analytic model. Patrick et al. observed that when there are less than 10 VoIP users in 802.11b, the quality of VoIP call is degraded seriously by performing experiment [6]. Kim et al. performed experiment of 802.11n in the presence of rate adaption [7]. However, this work only focused on the physical layer and did not consider the real-time application support in 802.11n. Lin et al. performed performance study of 802.11n supporting voice and video services [8]. However, the authors analyzed the performance based on proposed mathematical model.

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This paper investigates how actually the new techniques affect the performance of real-time applications. To this end, we measure delay, delay jitter, throughput, and packet loss rate of two applications (i.e. VoIP and video conference) with various numbers of data flows. In addition to experiments with static nodes, we also conduct experiments with human walking-speed mobility. For the first step of our project, initially we take channel bonding technique into account. Frame Aggregation and Reverse Direction techniques will be investigated in future work.

The rest of this paper is organized as follows. Section II briefly presents characteristics of real-time applications in terms of performance evaluation criteria. In Section III, we explore new key features of 802.11n (i.e. MIMO, Channel Bonding, Frame Aggregation, and Reverse Direction) that potentially improve performance of real-time applications. Section IV describes our experimental setup and also provides experiment results and the analysis. Finally, we conclude the paper in Section V.

II. REAL-TIME APPLICATIONS

This section briefly describes the characteristics of real-time applications that are different from conventional Internet services such as web browsing or file transfer. Real-time applications run on top of UDP transport protocol. They are sensitive to delay, delay jitter, and packet loss rather than throughput. They also frequently produce packets.

A. VoIP

Voice data traverses over IP networks and has a PSQM score equivalent or better to what would be seen over circuit switches. Delay, delay jitter, and packet loss in packets influence the quality of voice data. Long latency or delay is a nuisance on a two-way voice call. When one person speaks, there is a delay while the audio is transmitted to the distant end of the connection. This delay is seen again when the other person responds. High delay jitter on a connection can result in the audio sounding garbled if the audio packets are processed out of order or discarded all together. Minimal packet loss is not a problem as the human ear is not precise enough perceive the periods of missing audio. However, excessive packet loss can seriously impact voice quality.

B. Video Conference

High fidelity video is transmitted over IP with no degradation in quality. Like voice, video quality is impacted with the introduction of delay, delay jitter, and packet loss. Latency, if regular, isn't as much of an issue since video tends to be one way. In the case of two-way video conferencing, the same awkward delays would be present that one would experience in a high latency voice connection. Delay jitter on a video connection can quickly result in perceivable degradation of the video image. In fact, both delay jitter and packet loss will negatively impact the perceived quality of a video connection.

Table- I: Comparison of parameters between 20MHz and 40MHz channels

	20 MHz channel	40 MHz channel
Subcarrier Allocation	52 Data subcarrier 4 Pilot subcarrier 8 Guard tones	108 Data subcarrier 6 Pilot subcarrier 14 Guard tones
Symbol Time [ns]	50	25
Single Stream Rate [Mbps]	Long GI: 6.5 – 65 Short GI: 7.2– 72.2	Long GI: 13.5 – 135 Short GI: 15– 150

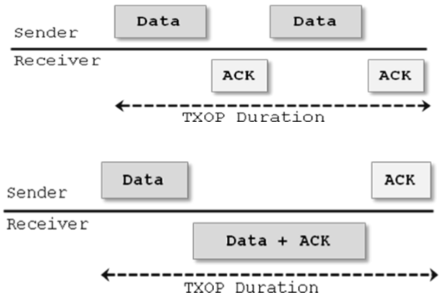

Fig. 1.IEEE 802.11n provides enhancement in MAC layer; transmission opportunity and reverse direction.

III. IEEE 802.11N ENHANCEMENT

A. Enhancement in Physical Layer

The enhancements in PHY layer of IEEE 802.11n over legacy WLANs are (i) multiple input and multiple output (MIMO) and (ii) channel bonding. In the legacy 802.11a/b/g WLAN system, a packet is transmitted by using one antenna. However, in 802.11n, a station with multiple antennas transmits and receives data packets using multiple antennas. This is called MIMO and MIMO technique is used for the wireless communication system where the transmitter and receiver are equipped with multiple antennas. By using MIMO technique, data rate and packet error rate can be enhanced.

In 802.11n, two adjacent channels, each of 20MHz are bonded to make 40MHz band. This is called channel bonding. When channel bonding is used, the symbol time is halved from 50ns to 25ns. Table I compares the parameters when channel bonding is not used to them when channel bonding is used. Since the bandwidth is enlarged when channel bonding is used, the transmission rate increases. Hence, the throughput and delay performance increase when channel bonding is used. In general, there are legacy WLAN system and interference in 2.4 GHz channel. Moreover, just 3 channels 1, 6, and, 11 is not overlapped. Hence, a channel bonding does not result in significant performance enhancement in 2.4 GHz. However, there are much more non-overlapping channels in 5 GHz. Hence, the channel bonding is more appropriate in 5 GHz frequency band than 2.4 GHz frequency band.

B. Enhancement in MAC Layer

The enhancements in MAC layer of IEEE 802.11n are (i) frame aggregation and (ii) reverse direction. Frame aggregation is devised for enhancing the efficiency and throughput.

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A frame aggregation significantly reduces MAC and PHY overhead. In 802.11n standard, two frame aggregation mechanisms are proposed, MAC Service Data Unit aggregation (A-MSDU) and MAC Protocol Data Unit aggregation (A-MPDU) [5]. A-MSDU aggregates multiple MSDU to form a large PSDU (PHY Service Data Unit). When the designated number of MSDUs arrives at the upper layer of MAC, the MAC framer generates PSDU by concatenating MSDUs and adding MAC header and frame check sequence (FCS). On the other hand, A-MPDU aggregates multiple MPDU to generate a large PHY Service Data Unit (PSDU) by concatenating several MPDUs with its own MPDU header and FCS. A delimiter is added between the MPDUs in order to indicating the start of each MPDU and padding bits are appended at the end of each MPDU.

Reverse Direction is devised to improve Quality of Service (QoS) support and overall efficiency. Fig. 1 shows the mechanism of reverse direction. Reverse Direction enhances transmission opportunity (TXOP) towards bi-directional traffic. To be specific, a reverse direction allows the station that accesses the channel in the remaining time in TXOP of its peer for reverse data flow. In order to using the remaining time, the duration of data flow should be smaller than the remaining time.

In this paper, there are many limitations on device settings, especially in MAC parameters, i.e. frame aggregation, reverse direction. Therefore, we only focus on the performance according to whether the effect of channel bonding is considered.

IV. EXPERIMENTS

This section describes our experiments and results. Among three properties analyzed in the previous sections, we conduct experiments with respect to the channel bonding option. The AP does not support the reverse direction option, and most off-the-shelf laptops used as clients in our experiments do not provide detailed specification of the communication chip. Thus, we cannot assure that all the clients execute the frame aggregation. Due to limitation of devices, we turn off these options in our experiment and remain further experiments on other two properties as future works.

A. Experiment Setup

For experiments, we build a network as shown in Fig. 2 at the 4th floor of Boelter Hall. The servers are connected to the Access Point (AP) through the Ethernet. Each server transmits and receives data packets to a corresponding client which connected to the AP via 802.11n channel. For traffic generation, we use both *iperf* [9] and *d-itg* [10]. The AP operates in 5 GHz frequency band, turns on and off the channel bonding option in order to evaluate its impact on real-time applications. Table II describes the experimental parameters used in our experiments. In order to avoid any interference with other systems running nearby areas, we select an appropriate channel after scanning wireless signals.

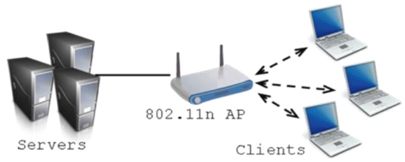

Fig. 2. **Servers are connected to the 802.11n AP through the Ethernet that forwards data packets to clients.**

Table- II: Parameters used in experiments

Configuration	*Setup*
Operating Frequency	5 GHz
Channel Bonding	Turn ON/OFF at AP
Frame Aggregation	OFF
Rate Adaptation	OFF
Reverse Direction	Not supported at AP
MIMO Option	Default (2x2)
Application	VoIP and video conferencing
Num. of Clients	2, 4, 6, 8, 10, 12
Mobility Support	8 static client + 4 mobile client

Table- III: Specification of voice codec (G.711)

Codec	*Bit Rate*	*Sample Period*	*Arrival Rate*	*Payload*
G.711.1	64 Kbps	10 ms	100 frames/s	80 bytes
G.711.2	64 Kbps	20 ms	50 frames/s	160 bytes

VoIP and video conferencing are considered as applications which run both on the servers and the clients. Note that the video conferencing application also generates bi-direction traffic on top of UDP. At each experiment, one server establishes a connection to one client, where one VoIP or video conferencing traffic is exchanged. During the experiments, we increase the number of clients from 2 to 12, all of which run the homogeneous application. In order to estimate influence of mobility on applications and 802.11n, we conduct additional experiments where 4 clients move around while 8 clients are stationary.

B. Performance of VoIP Application

In this experiment, each server and client establish a VoIP traffic connection via the 802.11n AP. The VoIP application uses the G.711 voice codec which is now widely used in many commercial products, which has a fixed bit rate of 64Kbps. In order to evaluate the impact of the sample rate of voice packets, we conduct two separate experiments with different sample rates; 10 ms and 20 ms. Table III shows specification of the voice codec used in the experiments. The G.711.1 codec generates a voice packet of 80 bytes every 10 ms. On the other hand, the G.711.2 codec generates a voice packet less frequently, i.e., every 20 ms.

A voice quality is typically measured as a Mean Opinion Score (MOS), which is a function of packet loss rate, delay, delay jitter, and codec-specific parameters [11]. The MOS value ranges from 1 to 5, where 5 represent the best quality. For comparison with a legacy 802.11b system, the MOS values shown in [1] are invited. Note that these were measured when a VoIP application used 10-ms G.729 voice codec. But, since the MOS represents a generalized number denoting the voice quality, it can allow us to compare performance roughly. In this experiment, we measure delay, delay jitter, packet loss rate, and throughput, and then calculate the MOS for both downstream (server to client) and upstream (client to server) of VoIP traffic.

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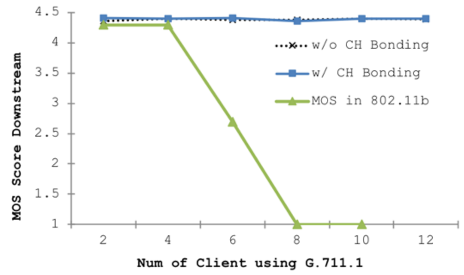
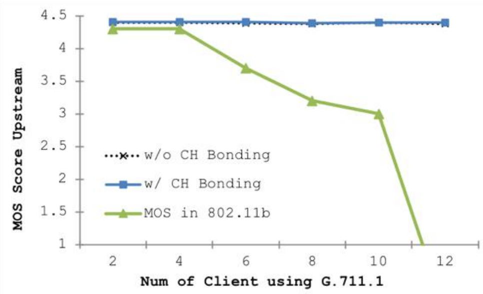


(a) MOS values for downstream traffic with increasing number of clients.



(b) MOS values for upstream traffic using G.711.1.

Fig. 3. **For voice quality, MOS values are calculated based on measurement for both downstream and upstream of VoIP traffic in 802.11n wireless channel.**

Fig. 3 depicts performance of VoIP application with varying number of clients in 802.11n system. An intuitive observation tells that the voice quality does not get worse, even when the number of clients increases up to 12. The MOS value maintains around 4.3~4.4 over all the scenarios. This is directly compared with the MOS values measured in 802.11b system, where the voice quality starts dropping when there are more than 6 clients. The main reason that makes difference, we believe, is the higher data rate provided in 802.11n. Faster transmission suppresses packet accumulation in a buffer, thus mitigates overhead due to contention. This reduces the packet drop rate which primarily determines the MOS value. Our analysis extends the result observed in [6], where 802.11g system supports high quality of VoIP applications with up to 8 clients. According to previous simulation study in [8], 802.11n can support more than 20 VoIP clients simultaneously when using 10-ms G.711 voice codec.

Due to high capability of 802.11n that can support at least 12 VoIP clients, Fig. 3 does not show any impact of the channel bonding property on the VoIP application. The VoIP traffic shows satisfactory performance regardless of the number of clients in our experiment. The high capability also prevents us from observing consequences of less-aggressive VoIP traffic having reduced sample period of 20ms. Despite limitation of our experiment, we believe that the channel bonding property would have an effect on the real-time applications. In order to verify our analysis, we conduct additional experiments with a video conferencing application which also generates bi-direction data traffic, but having much bigger payload size. It is expected that a large packet would easily saturate the wireless channel between the AP and the clients, i.e., the packet arrival rate becomes higher than the service rate, and thus the buffer accumulates packets. In this situation, we can easily see how the channel bonding property influences performance of real-time applications, which is examined in the following subsection.

Table- IV: Video codec (H.264/MPEG-4 AVC)

Level Number	*Bit Rate*	*Sample Period*	*Arrival Rate*	*Frame Payload*
1.3	768 *Kbps*	30 *ms*	100 *frames/s*	3200 *bytes*

C. Performance of Video Conferencing

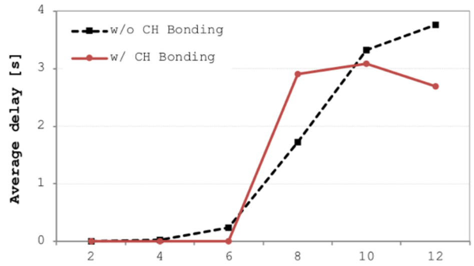
This subsection discusses experimental results of video conferencing applications on IEEE 802.11n. Our experiment uses the H.264/MPEG-4 AVC video codec whose specification is shown in Table IV.

Fig. 4 and Fig. 5 depict that the wireless network is saturated when there are more than 8 clients. The improvement of delay and delay jitter when a channel bonding is shown in Fig. 4. When a channel bonding is used, average delay and delay jitter is reduced by 27% and 50% on average, respectively. The reason of performance improvement is that a channel bonding reduces a symbol time. Hence, the unit time in 802.11n, i.e. Inter-Frame Space (IFS), back-off slot time, is halved. This results in the decrease of average delay and delay jitter. As shown in Fig. 4, when channel bonding is used, the packet drop rate increases by 21% and throughput is reduced by 11% on average. The reason of throughput enhancement is that a transmission rate increases when channel bonding is used. In addition, the reason of reduction in packet drop rate is that the increase in data transmission rate causes the decrease of the frequency of transmission attempts.

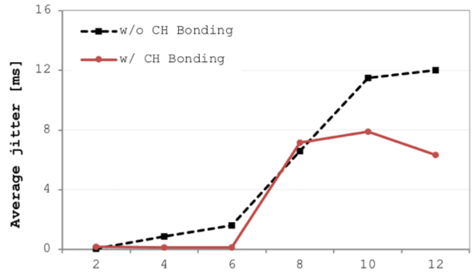
However, why only video streaming is saturated, not VoIP in our experiment? The data packet size of video streaming is 1080 Bytes, but VoIP is just 80 Bytes. The data packet size of video streaming is 13 times larger than that of VoIP. In wireless channel, the transmission time of video streaming is larger than that of VoIP since the packet size of video streaming is larger than that of VoIP. Hence, in video streaming case, wireless channel is saturated faster than the case of VoIP.

Fig. 5 shows the result of uplink video streaming. In the results of uplink video streaming, a channel bonding improves average delay and delay jitter. As can be seen in Figure 5, average delay and delay jitter is reduced by 47% and 65% on average, respectively. In addition, when channel bonding is used, we can see that packet drop rate and throughput are improved by 16% and 23%, respectively.

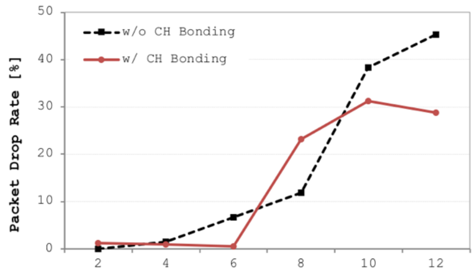
Why is the performance of uplink streaming better than the performance of downlink streaming? Especially, channel bonding can directly improve throughput, but not delay and packet drop rate. If using channel bonding, client is directly improved, because clients have each queue. However, AP is not directly improved, since the queue of AP share data transmitted from each server. Therefore, the improvement of AP is in inverse proportion to the number of flows.



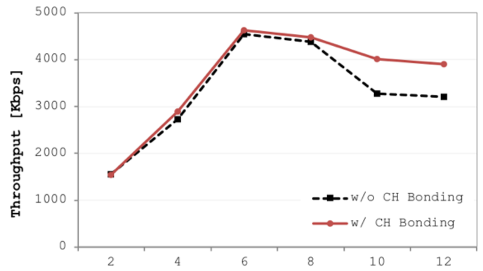
(a) Average delay



(b) Delay jitter

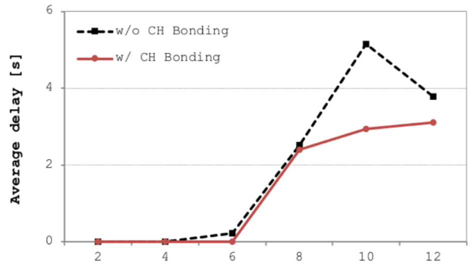


(c) Packet drop rate

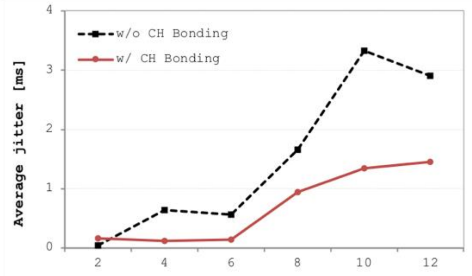


(d) Throughput

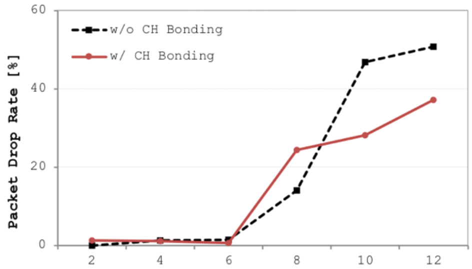
Fig. 4.**Measurement of network performance in downstream of video conferencing traffic. The X-axis represents the number of clients in all graphs.**



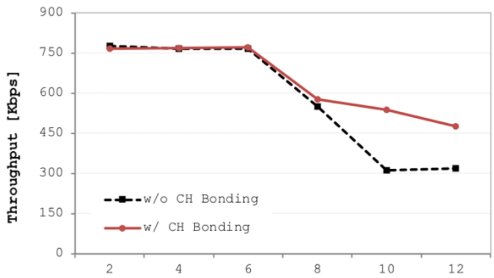
(a) Average delay



(b) Delay jitter



c) Packet drop rate



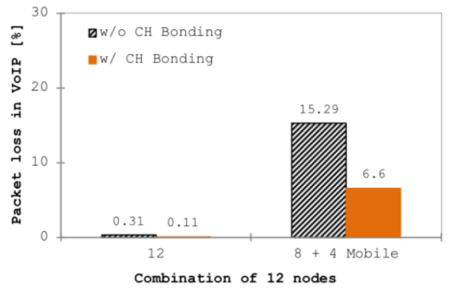
(d) Throughput

Fig. 5.**Measurement of network performance in upstream of video conferencing traffic. The X-axis represents the number of clients in all graphs.**

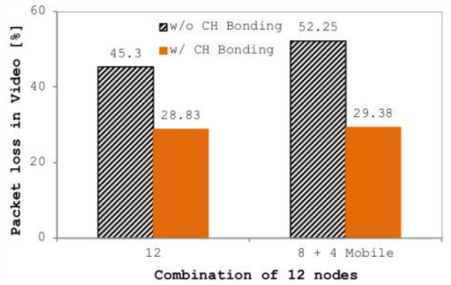
All measurement values are saturated when there are more than 8 clients. Under heavy contention, channel bonding affects delay, packet loss, and throughput. The channel bonding can improve the bandwidth from 20 MHz to 40 MHz. The contention is reduced due to wider bandwidth. Decreasing contention can improve delay, packet loss and throughput. In contrast, if contention is not heavy, channel bonding improves less than the case when the contention is heavy.

D. Impact of Channel Bonding on Mobility-Induced Packet Loss

Mobility on a wireless channel is known to degrade communication performance significantly. To evaluate its impact, our experiments record packet loss rate [%] in two scenarios; all 12 clients are stationary while a mobile scenario includes 4 mobile clients and 8 stationary clients. Experiment results confirm that mobility leads to increase of packet loss rate. In both VoIP and video conferencing applications, mobility of 4 clients produces additional packet loss of 8~15%.



(a) Packet loss rate in VoIP application



(b) Packet loss rate in video conferencing application

Fig. 6. **Packet loss rate of downstream traffic in VoIP and video conferencing applications; 4 clients out of 12 move around in the mobile scenario.**

Fig. 6(a) depicts that the packet loss rate grows from 0.31% to 15.29% in the case of mobile VoIP traffic, where the channel is not fully saturated. Considering that the packet loss rate mainly affects performance of real-time applications, it is worth investigating influence of the channel bonding property on mobility-induced packet loss. The figure also shows that the channel bonding property restrains the increase of packet loss in VoIP traffic. As a result, the MOS value reaches 3.75 in the mobile scenario, which is acceptable quality for voice communication while the MOS value drops down to 2.5 without the channel bonding option. In video conferencing traffic as shown in Fig. 6(b), the packet loss rate does not explicitly increase with additional mobility constraint while there is an additional increase of packet loss rate of 15% (from 45.3 to 52.25) without the channel bonding option. In this situation, the wireless channel is saturated, and overhead due to severe contention already contributes to huge loss of data packets even in static environment. Therefore, added mobility influences the packet loss rate less rigorously than the unsaturated situation. The channel bonding property mitigates the contention overhead and corresponding packet loss rate, and thus compensates the packet loss due to mobility.

V. CONCLUSION

IEEE 802.11 based WLAN technology is widely adopted and expected to the most popular access network in the future. This promising would come with recent amendment of 802.11n aiming at providing high throughput. In this study, we examined whether the new 802.11n standard can support emerging real-time applications such as VoIP and video conferencing, which were proved to make inefficient use of WLAN. We, first, analyzed three properties introduced by 802.11n (i.e., channel bonding, frame aggregation, and reverse direction) and expected their impact on the real-time applications. The experiment study, focused on the channel bonding property, showed that 802.11n improves real-time application performance slightly, but not satisfied. It could support at least 12 and 6 connections of VoIP and video conferencing applications, respectively. Nonetheless, this is a nontrivial improvement compared with legacy 802.11 systems.

Our analysis also demonstrated other potentials of 802.11n capable of supporting real-time applications. Yet, this analysis remains further experiment. Therefore, the future works will include more experiments on the frame aggregation and reverse direction with real-time applications.

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