

# Impact of Various VoIP and Video Traffics to Performance of Aggregation with Fragment Retransmission (AFR) in WLAN

Teuku Yuliar Arif and Riri Fitri Sari

Electrical Engineering Department, Engineering Faculty, University of Indonesia, Depok, Indonesia

## Summary

Aggregation With Fragment Retransmission (AFR) is a frame transmission scheme that can improve the efficiency and the throughput of MAC layer in WLAN 802.11n. In this paper, we studied the performance of AFR scheme when various VoIP and video traffics transmitted over WLAN. The purpose of this work is to know how the number of connections, packet size and fragment size of VoIP and video traffics influence the throughput and the delay performance of frame transmission. We used G.711, G.723a and G.729 format as VoIP traffics and H.264/MPEG-4 AVC format as video traffics which used 768 Kbps, 2 Mbps, 4 Mbps, 10 Mbps and 20 Mbps video bit rate. We used network simulator 2 (NS-2) with AFR module to simulate VoIP and video traffics over WLAN and measured each traffic performance. Simulation results show that throughput of MAC layer with AFR scheme will saturate when 70 connections of G.711 were transmitted. The transmission delay of G.723a and G.729 will be optimum when each traffic aggregated in 512 bytes of frame size. Simulation results also show that the throughput of AFR scheme is 70 Mbps and the delay is 0.025 ms when 30 H.264 video connections with 2 Mbps bit rate were transmitted. We have found the fragment and frame sizes that will give optimum throughput and the delay performance in AFR MAC scheme.

## Key words:

WLAN 802.11n, AFR, Performance, VoIP, Video

## 1. Introduction

Today, Wireless Local Area Network (WLAN) technology based on IEEE 802.11 standard has rapidly developed and widely deployed as wireless Internet access in office, home and public location. On the other hand, the requirement of multimedia applications such as Voice over IP (VoIP) and video also have improved which multimedia application needed to high bit rate and low latency. For example, VoIP application that used G.711, G.723a and G.729 codec format will need a low and constant latency [1,2,3], video H.264/MPEG-4 Advanced Video Coding (AVC) with DVD quality format will need to 9,8 Mbps bit rate and H.264 with HDTV quality format will need 20 Mbps bit rate [4,5].

The latest standard of WLAN is IEEE 802.11n released in late of 2009, it can support up to 600 Mbps data rate at PHY layer [6]. Certainly this standard is very potential to be used to transmit VoIP and video traffics. But

high data rate at PHY layer will not automatically improve MAC layer throughput because it depend on the mechanism in MAC layer itself. A research [7] shows that efficiency of IEEE 802.11 MAC layer will be decreased when PHY data rate increased. This occurred while PHY data rate increased causing faster transmission of MAC frame payload. Overhead such as PHY headers and contention time typically do not decrease at the same rate and thus begin to dominate frame transmission times.

To support the requirement of higher throughput MAC layer in WLAN IEEE 802.11n, Tianji Li *et al.* [7] introduced a new frame transmission mechanism called Aggregation With Fragment Retransmission (AFR). In AFR scheme, multiple packets received from the upper layer will be aggregated into a big frame before transmission. If the packet size is larger than fragment threshold size then the packet is divided into fragments before being aggregated. If the error occurred during transmission, then only the fragments of the frame that had been corrupted will be retransmitted. Therefore the AFR scheme can improve the throughput of MAC layer with small overhead.

Several papers reported the performance of MAC layer with AFR scheme and other schemes [8,9,10,11,12]. However the performance of AFR MAC mechanism when used to transmit VoIP and video traffic have not clearly described. For example for the standard format such as G.711, G.723a and G.729 in VoIP traffics format, and H.264/MPEG-4 AVC in video traffic format, with various bit rate, resolution size and the number of frame per second. It is important to know the behavior of the throughput and the delay of each VoIP and video traffic format while transmitting over WLAN with AFR MAC scheme. The transmission depends on the number of video connections, frame size and fragment size. The aims of this study are to gain the size of fragment and the size of the frame use by AFR scheme to get the optimal performance for each VoIP and video traffic format over WLAN.

The reminder of this paper is organized as follows. On Section II, we explain the AFR MAC scheme which include the aggregation scheme and the theoretical background of AFR performance. Section III describe the overview of VoIP and video traffics format, Section IV

presents detailed analysis of our simulation result. Finally we summarise our findings in Section V.

## 2. AFR Scheme

The main idea of AFR scheme is to aggregate several packets received at MAC layer from the upper layer into one large frame [7]. AFR scheme also limited the threshold of fragmentation size. If the size of a packet is larger than the size of the fragment, then the AFR scheme will divide the packet into several fragments.

Based on the aggregation and fragmentation mechanism in AFR scheme, MAC layer will transmit a large frame containing multiple fragments. Fragmentation will be used to retransmit only the corrupted fragments when data frame error detected at receiver. The illustration of the aggregation and fragmentation mechanism to VoIP and video packets in AFR scheme is shown in Fig. 1.

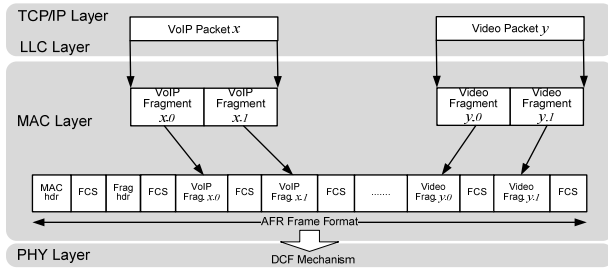


Figure 1. VoIP and Video Packet in AFR scheme

At the sender side, every packet will be segmented according the threshold of the fragment size. Before transmitted, each fragment will keep in sending queue (Sq). The MAC layer constructs a frame as follows: MAC layer searches the fragment in Sq and it aggregates the found fragments into a large frame until either no undelivered fragments are available or the frame size is large enough. Then, the MAC layer transmits this frame according to the procedure described in DCF mechanism.

The sender adds an FCS field for each fragment so that at the receiver side, all the FCS have to be checked for error detection. If all the fragments are received correctly, the receiver constructs a corresponding positive ACK message. If there are some fragments in error, then the receiver indicates the erroneous fragments in a bitmap field. All the successfully received packets are transferred to the upper layers and removed from the receiving queue.

Among the advantages of the AFR scheme is the suggestion of an adaptive waiting mechanism, in which the MAC layer never waits for packets to aggregate. In the case of high loaded network, a frame is retransmitted several times before being correctly received on account of collisions, every time a frame is retransmitted, the MAC

layer will be looking for more packets to fill the frame if it is not large enough.

If the channel is noisy, the MAC layer aggregates the available packets at the sending queue immediately after a failure transmission. In other case, in which the channel is not very noisy, AFR scheme behaves similarly to DCF function using the zero-waiting mechanism. Generally, the zero waiting mechanism improves the performance of AFR scheme in term of error rate.

In the AFR scheme, a MAC frame contains a frame header, fragment headers, several fragments and FCSS. The MAC header contains the same fields as in the DCF MAC header plus three new fields : the fragment sizes which represents the size of the fragment, the fragment number which represents the number of the fragments in the MAC, and a spare field. The AFR frame format is shown in Fig. 2.

Based on the theoretical analysis from [7], the saturation throughput ( $S_{AFR}$ ) of the AFR scheme is define :

$$S_{AFR} = \frac{P_3 \cdot L_f \cdot (1 - p_e^{frag})}{P_1 T_I + P_3 T_3 + P_C T_C} \quad (1)$$

where  $L_f$  is the length of a full frame and  $p_e^{frag}$  is the fragment error rate define from :

$$p_e^{frag} = 1 - (1 - p_b)^{L_{frag} + L_{FCS}} \quad (2)$$

$P_I, P_3, P_C, T_I, T_3, T_C$  are the probabilities and durations of Idle, Success/Error and Collision event respectively. The per packet MAC delay ( $D_{AFR}^{mac}$ ) is :

$$D_{AFR}^{mac} = r \cdot \frac{P_3 \cdot T_I + P_3 \cdot T_3 + P_C \cdot T_C}{P_3} \quad (3)$$

where  $r$  is the expected number of retransmission attempts in which :

$$r = \sum_{r'=1}^{\infty} r' \left[ \left( 1 - (p_e^{frag})^{r'} \right)^{m'} - \left( 1 - (p_e^{frag})^{r'-1} \right)^{m'} \right] \quad (4)$$

Where  $m'$  is the number of fragments in a packet and  $r'$  is the probability of retransmit a fragment. The reader who interested with the detail explanation of all above equation can refer to [7].

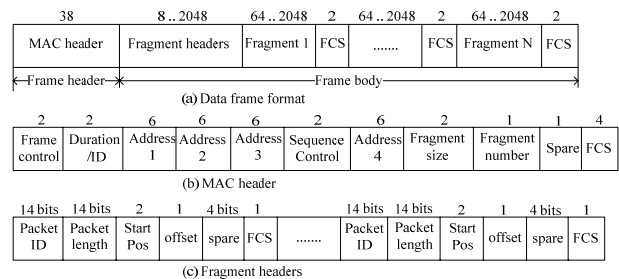


Figure 2. Data format in AFR scheme

### 3. VoIP and Video Traffic Evaluation

In this paper we evaluated the performance of MAC layer of WLAN IEEE 802.11n which use AFR scheme while transmitting VoIP and video traffic. Generally, in a VoIP or video transmission system, analogue signals are first digitized, compressed and encoded into digital voice or video streams by the codecs. The output streams are then packetized for efficient and network friendly transmissions over an IP-based network [10, 11]. In general, multimedia streams are encapsulated with RTP/UDP/IP headers. After the voice or video packets are delivered through the network, the reverse processes of decoding and depacketizing are conducted at the receiver.

#### 3.1 Evaluation of VoIP traffic

The main attributes of some frequently used voice codecs are listed in Table 1. Different codecs use different compression algorithms resulting in different bit rates. G.711 is the international standard for encoding telephone audio, which has a fixed bit rate of 64 Kbps. With a 10 ms sample period, corresponding to a rate of 100 packets per second, the payload size is  $64000/(100 * 8) = 80$  bytes. When the sample period is increased to 30 ms, corresponding to 33.33 packets per second, the payload size is increased to 240 bytes accordingly. G.723 and G.729 have a lower bit rates and higher codec complexity compared to G.711. G.723 is one of the most efficient codecs with the highest compression ratio and G.729 is an industry standard with high bandwidth utilization.

Table 1. Specification of voice codec

Voice codec		G.711	G.723a	G.729
Codec bit rate (Kbps)		64	5.3/6.3	8
Sample Period (ms)	Arrival rate (fps)	Payload (bytes)	Payload (bytes)	Payload (bytes)
10	100	80	-	10
20	50	160	-	20
30	33.33	240	20/24	30
40	25	320	-	40
50	20	400	-	50

#### 3.2 Evaluation of Video traffic

Wireless video streaming service is another promising and demanding service in the next generation WLANs. Some video-based applications include video telephony, video conferencing, IPTV, etc. There are a large number of media platforms for video services, the majority of which employ the ITU-T H.26x video standards, including H.261, H.263, and H.263+, etc. H.264/MPEG-4 AVC is one of the latest international video coding standard that supports very high data compression. The H.264 codec has a broad range of applications that covers all forms of digital video from low

rate Internet streaming applications (e.g., 64 Kbps) to broadband high definition video (HDV) applications (e.g., 240+ Mbps).

Two main objectives of H.264 video coding are to enhance the coding efficiency and improve the network adaptation. H.264 codec consists of two conceptual layers, Video Coding Layer (VCL) and Network Abstraction Layer (NAL) [14]. The VCL contains the signal processing functionality of the codec such as transform, quantization, motion search/compensation, and the loop filter, and outputs video slices. The NAL encapsulates the slices into NAL units (NALUs), which are suitable for transmission over packet networks.

In the standard, levels specify the maximum frame size in terms of the total number of pixels/frame. H.264/MPEG-4 AVC defines 16 different levels, tied mainly to the picture size and frame rate [5]. Some examples for various resolution, frame rate, and maximum compressed video rate in five levels are listed in Table 2 that will use for video traffics evaluation over AFR scheme in this paper. At a particular level, if the picture size is smaller than the typical picture size, then the frame rate can be higher than the typical rate. For example, the level 2 supports up to 2 Mbps video rate, with the frame rate of 30 frames per second (fps) at the frame resolution of 320 x 240 pixels, and level 3 supports up to 10 Mbps video rate with the frame rate of 30 fps at the resolution of 720 x 480 pixels. Higher resolution provides better image quality and higher frame rate results in a smoother motion video.

Table 2. Levels on H.264/MPEG-4 AVC

Level Number	Video bit rate (bps)	Resolution @ frame rate (fps)
1.3	768 k	352 x 288 & 30
2	2 M	352 x 288 & 30
2.2	4 M	720 x 480 & 15
3	10 M	720 x 480 & 30
3.2	20 M	1280 x 1024 & 42

### 4. Simulation and Result

We simulated a simple network model to evaluate how AFR scheme influences the performance of VoIP and video traffic over WLAN. The WLAN consists of an Access Point (AP), and the AP will be connected to several wireless stations (STAs). AP also assumed connected to multimedia servers which have VoIP and video to be services. Therefore in our network model, several STAs will be connected to several multimedia servers.

To simulate that network schenario, we use network simulator 2 (NS-2) with an AFR module from Hamilton Institute, Ireland [17]. Table 3 shows the parameters we

use for VoIP traffics simulation, where as Table 4 shows the parameters we used for video traffic simulation.

Fig. 3 shows the impact of the number of VoIP connections to MAC layer throughput with AFR scheme. In our simulation, we are use VoIP traffic that used G.711, G.723a and G.729 format in which each payload size are 240, 24 and 30 bytes, and each bit rate are 64 Kbps, 6.3 Kbps and 8 Kbps. Every traffic has a sample period at 30 ms and the arrival rate is 33.33 fps as shown in Table 1. Each payload is encapsulated with 40 bytes RTP/UDP/IP header to make a packet with the size of 280 bytes, 64 bytes and 70 bytes. In AFR scheme, G.711 packet will be divided into fragments that has 64 bytes or 128 bytes size. On the other hand, G.723a and G.729 will be divided into fragments with only in 64 bytes size.

The simulation result shows that the throughput of AFR MAC will be increased if the number of VoIP connection is increased too. Throughput of G.711 is higher than G.723a and G.729 because G.711 is used for VoIP bit rate higher than G.723a and G.729 bit rate. In VoIP simulation parameters, when we used 6 Mbps AFR data rate and 1 Mbps basic rate, the throughput of G.711 will saturate at 1.9 Mbps in which the number of VoIP connection are 70 users. However this is not the case with G.723a and G.729 which only have 0.2 Mbps MAC throughput. The throughput showed in Fig. 3 occurred when the channel Bit Error Rate (BER) is  $10^{-4}$  and AFR frame size is 1024 bytes.

Fig. 4 and 5 shows the average delay of G.711, G.723a and G.729 based on the aggregated frame size, different BER condition, and the number of VoIP connections are 90 users. G.711 with 64 bytes fragment size can use a frame with size from 512 bytes to 4096 bytes, and the size of 8192 bytes only can used by 128 bytes fragment size. On the other hand, G.723a and G.729 can use 64 bytes fragment size and use the frame size from 256 to 4096 bytes. This condition is caused by AFR scheme which was limited the maximum fragments in a frame is 64 units. The delay of G.711 will be optimum at 0.36 ms when it is used for 1024 frame size, 128 fragment size and the channel BER is  $10^{-6}$ . The average delay of G.723a and G.729 will be optimum at 0.003 ms when they use 512 bytes frame size.

Table 3. Parameters for VoIP simulation

	Fig. 3	Fig. 4	Fig. 9
Number of STAs	Varied	90	90
Data rate (Mbps)	6	6	6
Basic rate (Mbps)	1	1	1
BER	$10^{-4}$	$10^{-4,-5,-6}$	$10^{-4,-5,-6}$
AFR sending queue (packets)	10	10	10
AFR IFQ (packets)	10	10	10
AFR frame (bytes)	1024	Varied	Varied
AFR fragment (bytes)	64, 128	64, 128	64

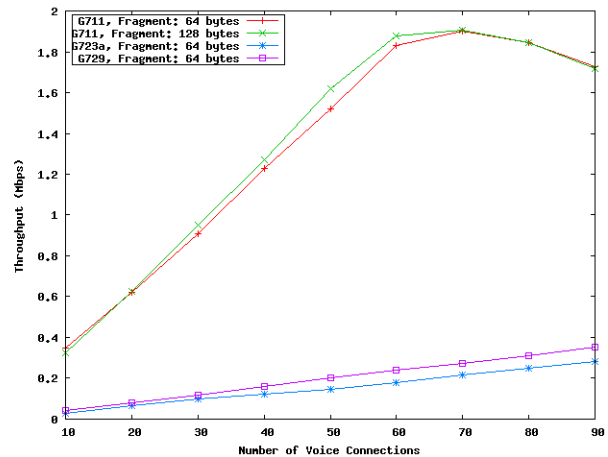


Figure 3. G.711, G.723a and G.729 throughput

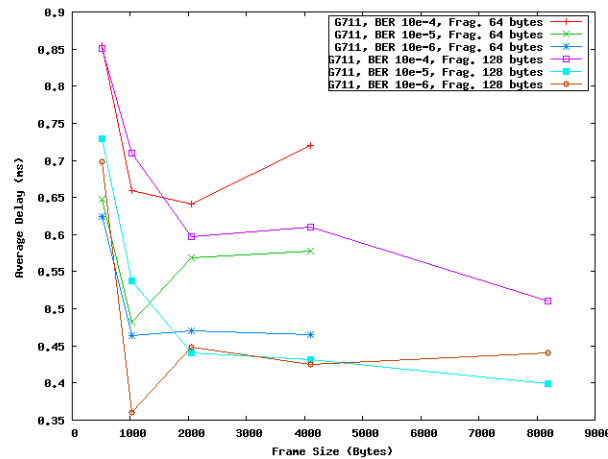


Figure 4. G.711 average delay

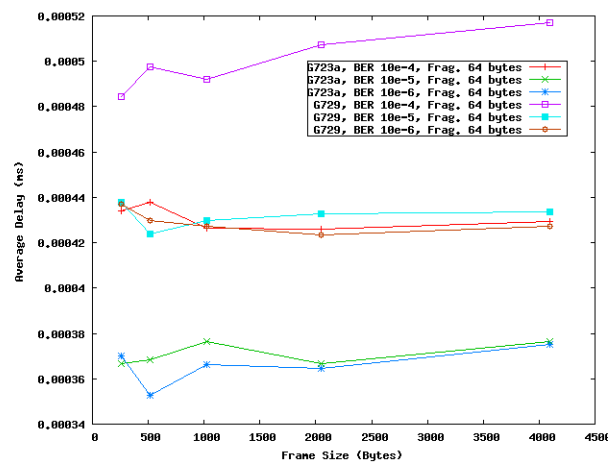


Figure 5. G.723a and G.729 average delay

Table 4. Parameters for video traffic simulation

	Fig. 6, 9	Fig. 7, 10	Fig. 8, 11
Number of STAs	Varied	30	30
Data rate (Mbps)	216	216	216
Basic rate (Mbps)	54	54	54
BER	$10^{-5}$	$10^{-5}$	$10^{-5}$
AFR sending queue (packets)	10	200	200
AFR IFQ (packets)	10	100	100
Packet (bytes)	1500	2048	1024
AFR frame (bytes)	9000	Varied	8192
AFR fragment (bytes)	750	512	Varied

Fig. 6, 7 and 8 show the impact of the number of H.264 video connection, frame size and fragment size to AFR MAC layer throughput. Parameters used in video traffic simulation is shown in Table 4. In this simulation we use H.264 video format with the bit rate are 768 Kbps, 2, 4, 10 and 20 Mbps. In Fig. 6, the packet size of each video traffic is 1500 bytes. Each packet is divided into some fragment by AFR scheme. The fragment size is 750 bytes and frame size is 9000 bytes. We set the data rate to 216 Mbps, basic rate to 54 Mbps, and channel BER is  $10^{-5}$ . Simulation results show the H.264 video traffic with 768 Kbps bit rate and 352x288 pixel resolution will increased the throughput of MAC AFR caused by increasing the number of video connection. For example, if number of video connections are 80, the throughput of MAC AFR will be 50 Mbps. If there are 10 H.264 video connections and 768 Kbps video bit rate, the throughput will be 9 Mbps.

The frame size will be significantly impacted the throughput of H.264 video transmission over MAC AFR, while video bit rate are 4 Mbps, 10 Mbps and 20 Mbps. On the other hand, the frame size will not significantly impact video traffic with 768 Kbps and 2 Mbps bit rate. Each video bit rate has 23 Mbps and 60 Mbps throughput while the number of video connection is 30 and the channel BER is  $10^{-5}$ . The optimal frame size for 4 Mbps and 10 Mbps video bit rate is 8192 bytes in which each has 92 Mbps and 85 Mbps throughput. The optimal frame size for 20 Mbps video bit rate is 4096 bytes as shown in Fig. 7.

The impact of the fragment size for every H.264 video traffic to MAC AFR throughput is shown in Fig. 8. In this simulation the number of video connections is 30, the packet size is 1024 bytes, the frame size is 8192 bytes and channel BER is  $10^{-5}$ . The fragment size used by each video traffic are 128, 256, 512 and 1024 bytes. In general, the fragment size of video traffic has no significant impact to the throughput of each video bit rate. In different fragment size, H.264 video traffic which used 768 Kbps and 2 Mbps bit rate will get the throughput of 23 Mbps and 56 Mbps. The maximum throughput of H.264 video traffic with 4 Mbps bit rate is 87 Mbps while the fragment size is 512 bytes. On the other hand, H.264 video traffic with 10 Mbps and 20 Mbps bit rate that used fragment size

256 bytes will get the maximum throughput at 84 Mbps and 80 Mbps.

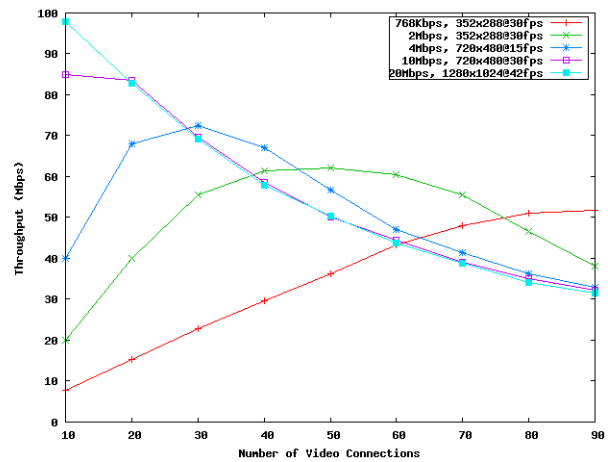


Figure 6. Throughput of varied number of video connections

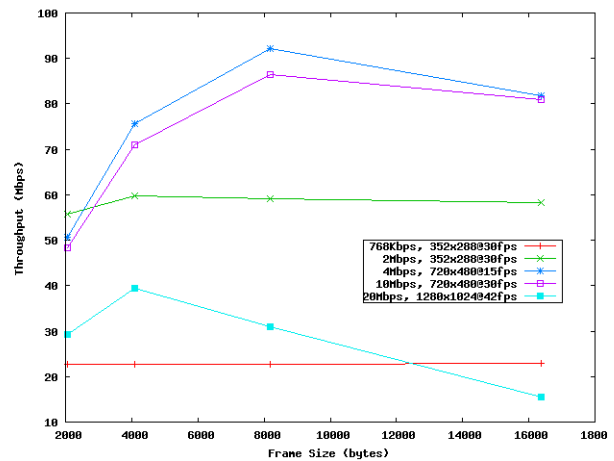


Figure 7. Throughput of varied frame sizes of video traffic

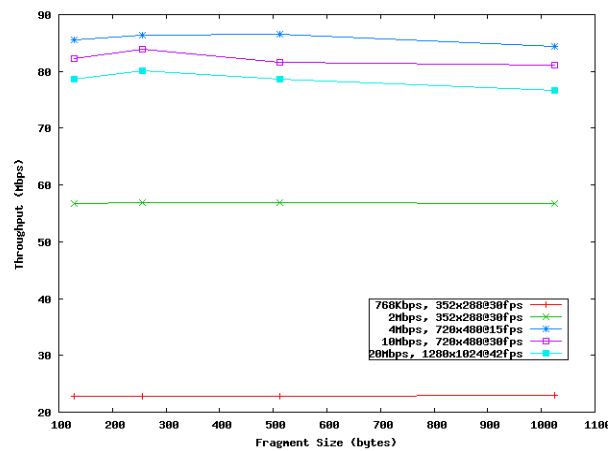


Figure 8. Throughput of varied fragment size of video traffic

Fig. 9, 10 and 11 depicted the number of the delay of H.264 video traffic based on the increased number of video connections, aggregate frame size, and fragment size. Parameter used in this simulation is shown in Table 4. Each graph that shown in Fig. 9 used the packet size of 1500 bytes. The packet is divided into the fragment size of 750 bytes and aggregated in a transmission frame with the size of 9000 bytes. In this simulation, we also used the data rate of 216 Mbps, the basic rate of 54 Mbps and channel BER is  $10^{-5}$ .

The simulation result shows H.264 video traffic with bit rate 768 Kbps and video resolution 352x288 pixel will cause the peak delay of MAC AFR to increase significant in line with the increase of the number of video connections. For example, if the number of video connection is 80, the MAC AFR delay will be 0.1 ms. The peak delay of video transmission will increase if we use the higher video bit rate. For H.264 video traffic with 10 Mbps bit rate and 30 video connections, the peak delay of video will be 0.3 ms.

Different frame sizes of video transmission will significantly affect video traffic with the bit rate of 4 Mbps, 10 Mbps and 20 Mbps in which the peak delay of transmission will increase if frame size of the video transmission also increased. The peak delay of 30 video connections with 2048 bytes packet size is shown in Fig. 10. The optimal frame size to get the lowest peak delay for video transmission with the bit rate of 2 Mbps, 4 Mbps, 10 Mbps and 20 Mbps is 16384 bytes. If we use 2048 bytes frame size or the same packet size, the peak delay will increase to 80%. The frame size will have no impact to H.264 video transmission peak delay with the bit rate of 768 Kbps and the average delay is below 0.2 ms.

The impact of fragment size of each H.264 video traffic to peak delay of MAC AFR is shown in Fig. 11. The parameters used in the simulation are 30 video connections, 2048 bytes packet size, 8192 bytes frame size and BER  $10^{-5}$ . The fragment size that can used by each video bit rate are 128, 256, 512 and 1024 bytes. In general, the video traffic fragment size will not impact significantly the peak delay of each video bit rate. For each fragment size, H.264 video with bit rate of 768 Kbps and 2 Mbps will get the peak delay of 0.1 ms and 0.4 ms. The peak delay of H.264 video traffic with bit rate 4 Mbps will be optimum if 512 bytes fragment size is used. It will get a peak delay of 1 ms. The peak delay of H.264 video traffic with 10 Mbps bit rate will be optimum at 1,2 ms with 128 bytes fragment size and the peak delay of H.264 video traffic with bit rate of 20 Mbps will be optimum at 1,4 ms with 1024 bytes fragment size.

Based on the analysis of the simulation result, we get the fragment size and frame size for all VoIP and Video traffics format which can give optimal throughput and optimal delay for each traffic as shown in Table 5.

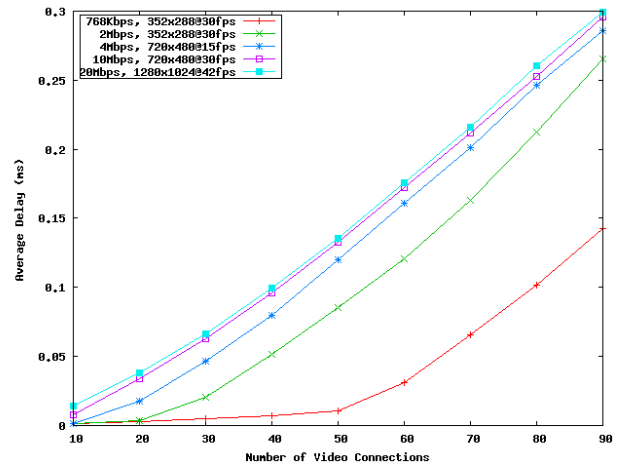


Figure 9. The average delay of varied number of video connections

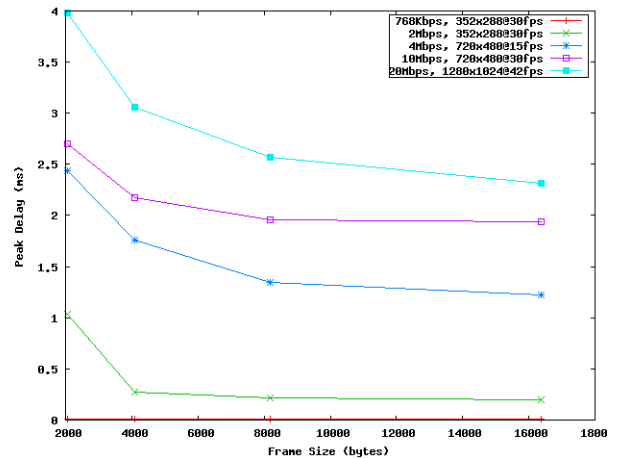


Figure 10. The peak delay of varied frame sizes of video traffic

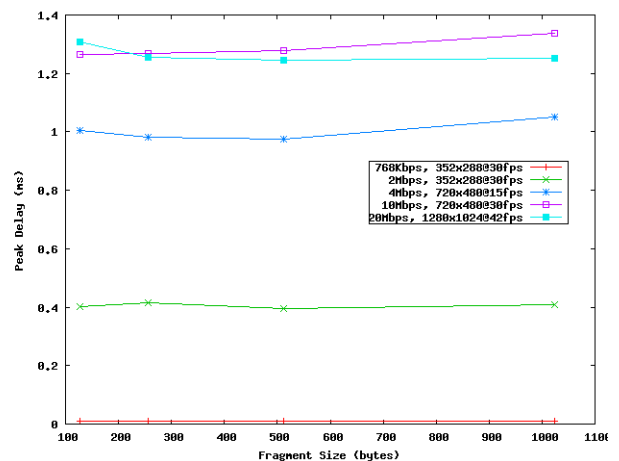


Figure 11. The peak delay of varied fragment sizes of video traffic

Table 5. The optimum parameter for AFR Scheme

VoIP/ Video	Rate (bps)	Optimal throughput		Optimal delay	
		Frag. size (bytes)	Frame size (bytes)	Frag. size (bytes)	Frame size (bytes)
G.711	64 K	128	1024	128	1024
G.723a	6.3 K	64	512	64	512
G.729	8 K	64	512	64	512
H.264	768 K	128	2048	128	2048
H.264	2 M	512	4096	512	16384
H.264	4 M	512	8192	512	16384
H.264	10 M	256	8192	512	16384
H.264	20 M	256	4096	128	16384

## 5. Conclusion

This paper has evaluated the impact of various VoIP dan video traffics to performance of AFR scheme which used in highspeed Wireless LAN. Simulation results have shown the VoIP traffic which use G.711, G.723a, and G.729 codec format will have different impact to MAC throughput and and frame delay of AFR scheme. The different impact depends on frame size and fragment size use in transmission of packets. Simulation results have also shown the video traffic which use various bit rates and various video resolution of H.264/MPEG-4 video format will generate different impact to MAC throughput and and frame delay of AFR scheme. From our simulation results analysis, we have found the optimum frame size and fragment size of each VoIP and video traffic which must be used by AFR scheme to get the optimal performance. In the future work we plant to improve the AFR scheme with QoS implementation.

## References

- [1] ITU-T, Recommendation G.711, <http://www.itu.int/rec/T-REC-G.711/e>, last access 10 June 2010.
- [2] ITU-T, Recommendation G.723, <http://www.itu.int/rec/T-REC-G.723/e>, last access 10 June 2010.
- [3] ITU-T, Recommendation G.729, <http://www.itu.int/rec/T-REC-G.729/e>, last access 10 June 2010.
- [4] ITU-T Rec. H.264 and ISO/IEC 14496-10 (MPEG4-AVC), "Advanced Video Coding for Generic Audiovisual Services", v1, May, 2003; v2, Jan. 2004; v3 (with FRExt), Sept. 2004; v4, July 2005.
- [5] Detlev Marpe, Thomas Wiegand, Gary J. Sullivan, "The H.264/MPEG4 Advanced Video Coding Standard and its Applications", IEEE Communications Magazine, August 2006.
- [6] IEEE 802.11n-2009-Amendment 5: Enhancements for Higher Throughput, 29 October 2009.
- [7] Tianji Li, Qiang Ni, David Malone, Douglas Leith, Yang Xiao, and Thierry Turletti, "Aggregation with Fragment Retransmission for Very High-Speed WLANs", IEEE/ACM Transactions on Networking, Vol. 17, No. 2, April 2009.
- [8] S. Maaroufi, W. Ajib and H. Elbiaze, "Performance Evaluation of New MAC Mechanisms for IEEE 802.11n", in Proc. IEEE Global Information Infrastructure Symp. (IEEE GIIS 2007), Marrakech, Morocco, 2-6 July 2007.
- [9] Alina Olteanu and Yang Xiao, "Security Overhead and Performance for Aggregation with Fragment Retransmission (AFR) in Very High-Speed Wireless 802.11 LANs", IEEE Transactions on Wireless Communications, Vol.9, No. 1, January 2010.
- [10] L. X. Cai, X. Ling, X. Shen, J. W. Mark, and L. Cai, "Supporting Voice and Video Applications over IEEE 802.11n WLANs", *ACM Wireless Networks (WINET)*, vol. 15, no. 4, pp. 443-454, May, 2009.
- [11] S. Vijay Bhanu, R. M. Chandrasekaran, V. Balakrishnan, "Effective Bandwidth Utilization in IEEE802.11 for VOIP", International Journal of Computer Science and Information Security, Vol. 8, No1, April 2010
- [12] Yaw-Wen Kuo, Tsern-Huei Lee, Yu-Wen Huang, Jing-Rong Hsieh, "Design and Evaluation of a High Throughput MAC with QoS Guarantee for Wireless LAN", Proceedings of the 2009 IEEE 9th Malaysia International Conference on Communications, 15-17 December 2009, Kuala Lumpur, Malaysia.
- [13] Yu-Tzu Huang, Jia-Shi Lin, Kai-Ten Feng, "Performance Analysis for Aggregated Selective Repeat ARQ Scheme in IEEE 802.11n Networks", IEEE 20th International Symposium on Personal, Indoor and Mobile Radio Communications, pp: 37, Tokyo, April 2010.
- [14] Wang and H. Wei, "IEEE 802.11n MAC Enhancement and Performance Evaluation," *ACM/Springer Mobile Networks and Applications Journal (MONET)*, Volume 14, Issue 6, Page 760-771, Dec. 2009.
- [15] Lin Y. and Wong VWS, "Frame aggregation and optimal frame size adaptation for IEEE 802.11n WLANs", Proceedings of IEEE Global Telecommunications Conference (GLOBECOM), San Fransisco, CA, U.S.A., November 2006.
- [16] NS-2, <http://www.isi.edu/nsnam/ns/>, last access 24 April 2010.
- [17] AFR Implementation, [http://www.hamilton.ie/tianji\\_li/afr.html](http://www.hamilton.ie/tianji_li/afr.html), last access 3 Mai 2010.