Comparative Analysis of IEEE 802.11g Multimedia Multicast Performance Using RTP with an Implemented Test-bed

Muhammad Faheem Mohd Ezani
Wireless Communication Cluster,
MIMOS Berhad,
Kuala Lumpur, Malaysia
faheem.ezani@mimos.my

Al-Sakib Khan Pathan
Department of Computer Science,
International Islamic University
Malaysia (IIUM), Malaysia
sakib@iium.edu.my

Shariq Haseeb
Wireless Communication Cluster,
MIMOS Berhad,
Kuala Lumpur, Malaysia
shariq.haseeb@mimos.my

Abstract— **In network layer protocols, the most efficient way of sending similar packets to a group of nodes is via multicast. Multicast is a profound concept that has been around for some time and has seen through its implementation in various aspects of today's network access technology. The IEEE 802.11 wireless standards, however, loosely honor IP layer multicast packets by encapsulating them in broadcast frames. This in turn degrades the 802.11 network access bandwidth capacity and further decreases its data transmission reliability. An enhancement workaround which has been devised was to encapsulate multicast packets in a unicast MAC layer frame. In addition, RTP was also observed to be an enabler to better user experience in multicast streaming. In this paper, we investigate the 802.11g multicast performance with and without the enhanced multicast mechanism. We further expand our investigation by implementing RTP during the multimedia multicast stream and without it. Our results lead us to conclude that there are significant advantages of using both the enhanced multicast mechanism and RTP during a multicast streaming via the 802.11g wireless standard.**

Keywords - Multicast, IEEE 802.11, Wireless, Multicast-in-MAC-Layer-Unicast, Performance, Reliability

I. INTRODUCTION

Network layer multicast is an efficient way of delivering data stream in a one-to-many connection. When a source node intends to send duplicated data stream to a set of receivers, only one data stream is required to be sent across the network before it is duplicated separately to the receivers' sub-network and eventually to the receivers in the sub-network themselves. Such mechanism is achieved through protocols that listen for multicast subscribers on the network (e.g., MLDv1 and MLDv2) and also those that establish the multicast routing path tree amongst the network's intermediate routers (e.g., PIM-SM, PIM-DM, and PIM-SSM). Assuming an all-wired network connection, achieving an end-to-end multicast communication rarely causes instability. The wide spur of wireless IEEE 802.11 access devices, however, introduce a data stream reliability concern especially when multicast packets are being transmitted.

Apart from this, the issue of limited user capacity on a given 802.11 [1] compliant access point is also of concern. When two users are connected to a wireless 802.11 access point, both users typically have to share the bandwidth. Theoretically, an 802.11g access point would have 54Mbps worth of throughput. Therefore, when two users are connected, both users would get approximately 27Mbps throughput. If the number of users increase, the throughput for each user would also throttle down and may have significant effect on the user experience. Several studies [9], [10], [11], [12] assert that the 802.11 protocol can be made to achieve certain maximal throughputs via different network device configurations. Our experimental setup aims to investigate throughput performance which can be achieved via the normal 802.11standards and a multicast-in-MAC-unicast mechanism. The multicast-in-MAC-unicast mechanism is a workaround to enhance multicast transmission over 802.11 and will later be explained in this paper. Alongside this, we also observed multicast performance where UDP and RTP/UDP protocols were used alternately to prove that the latter can further enhance the streaming performance.

Following the introductory texts, the remainder of this paper is organized as follows: Section II briefly discusses related works pertaining to multicast over 802.11a/b/g/n and their enhancements, if any. Section III expounds the technicalities of how multicast is traditionally transmitted over 802.11 and how the multicast-in-MAC-unicast mechanism is done to achieve improved throughput. In Section IV, we show our experimental setup and explain the parameters involved to measure data transmission reliability and multicast performance. Section V discusses the results obtained and analyzes them according to the parameters identified earlier. Finally, Section VI concludes our findings and suggests future research works.

II. RELATED WORKS

Many studies in [1], [2], [3], [4], [5], [6] were conducted to analyze multicast performance on 802.11 MAC protocols and suggest methods of enhancing its multicast transmission quality. Generally, existing solutions for 802.11 multicast packet transmissions address a common issue occurring at the MAC layer. According to the standards defined in 802.11, multicast packets are dictated to be sent in a MAC broadcast frame [7]. This in turn affects the transmission rate because a broadcast frame in 802.11 is always sent at a base data rate. On an 802.11b/g compliant access point, this would mean a mandated transmission rate of 1Mbps and a lack of feedback mechanism. The same applies to 802.11a whereby its base data rate is 6Mbps. In an effort to overcome this problem, Tanigawa et al. [8] took advantage of this fact and devised a translation mechanism in which network layer multicast packets are sent to receivers via a MAC unicast frame. This, in effect, forces the MAC layer to transmit the multicast data at the defined unicast rate in 802.11 and hence improves the data transmission performance.

Although having multicast packets sent or encapsulated in a unicast MAC layer frame may improve a multicast stream's end-to-end transmission, the overall user experience may still be compromised due to usage of the User Datagram Protocol (UDP). UDP transmissions alone are unreliable; they are prone to packet loss and out-of-sequence packet delivery [14]. Regardless, UDP is the preferred implementation for live multimedia streaming because it requires no packet acknowledgments and therefore minimizes transmission overheads. According to a simulation study in [15], UDP manifested as the least reliable transport protocol because it resulted in higher packet loss and lesser data throughput. The study further showed that there is a significant performance improvement when different MAC/PHY layer and transport layer protocol combinations are used for streaming. The reason for this is that if reliability can be enhanced from the lower layers, then any reliability-scheme implemented on top of those layers will theoretically enhance the overall data throughput reliability.

Since UDP has been designed without a feedback mechanism, the IETF has introduced the Real-time Transport Protocol (RTP) in order to provide better means of audio/video transmission [17]. One of the key functionalities of the RTP is its ability to rearrange blocks of data and identify packet loss during a streaming session. This is done through the RTP's timing and sequence number information contained within its header. In [18], a performance analysis was done between RTP and Skype's proprietary protocol on voice over IP (VoIP). This experiment used the Internet as part of its network testbed and was therefore a cable/wired connection setup.

Besides UDP, other transport protocols which have been discussed in [15], [16] are: TCP Forward Acknowledgement (FACK), TCP Selective Acknowledgement (SACK), TCP Reno, NewReno, and TCP Tahoe. These protocols are used to enhance the current implementation of Transmission Control Protocol (TCP) and are beyond the discussion of our paper. However, it should be noted that TCP, unlike UDP, strives for data reliability rather than timeliness. Real-time applications such as Voice over IP (VoIP) and IPTV benefits more from RTP which runs on UDP since it can provide essential parameters to support both timeliness and packet retransmission [19].

III. Our Experimental Setup

For our experiment, we wanted to observe multicast streaming performance over 802.11g connection using RTP/UDP and UDP only protocols. Our experiment also included observation of the performance when the 802.11 multicast enhancement mechanism is being used.

A. Network Testbed

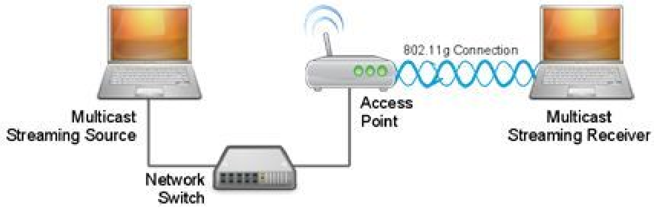

Figure 1. Network testbed experimental setup

The testbed network setup is illustrated in Figure 1. The Multicast Streaming Source was installed with VLC version 0.8.6. An ethernet cable was directly connected to a Network Switch. Wireshark [20] was also installed on the Multicast Streaming Source to monitor packets going out from the ethernet interface.

We used the MIMOS WiWi™ Outdoor Access Point, a locally developed access point supporting the standard 802.11b/g and our own multicast-in-MAC-unicast solution. The said solution is configurable whereby it can be enabled and disabled via a web interface. For our experiment, we configured the access point to operate only on 802.11g.

As per the Multicast Streaming Receiver, we installed VLC version 0.8.6 as the streaming receiver application. Wireshark was also installed to monitor any incoming packets on the 802.11g interface.

B. Performance Parameter

In order to measure the performance of the multicast packets being streamed, we monitored the number of packet loss incurred during the operation. Wireshark was setup on both the streaming source and receiver, and a comparison was being made based on the packets sent and received.

C. Methodology

Since our experiment was conducted in an office environment which also has other 802.11 access points in the same room, we ran each set of experiments up to ten times to observe the consistency of the data. The experiments conducted were as follow:

i. UDP streaming over standard 802.11g
ii. UDP streaming over 802.11g with enhanced multicast mechanism
iii. RTP/UDP streaming over standard 802.11g
iv. RTP/UDP streaming over 802.11g with enhanced multicast mechanism

As explained earlier, for each of the above experiments, we executed ten times. For each execution, we captured the number of packets sent from the streaming source and those received at the receiving node. We used the same video for each execution and the same streaming bit rate which is averaged at approximately 796kbps.

IV. Results and Analysis

The following section discusses the results gained from our experiments as mentioned in the previous section.

A. UDP Streaming over Standard 802.11g

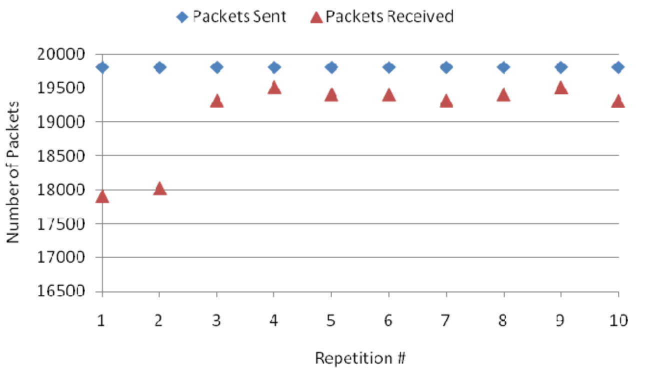

Figure 2. Number of packets sent and received via UDP streaming over standard 802.11g

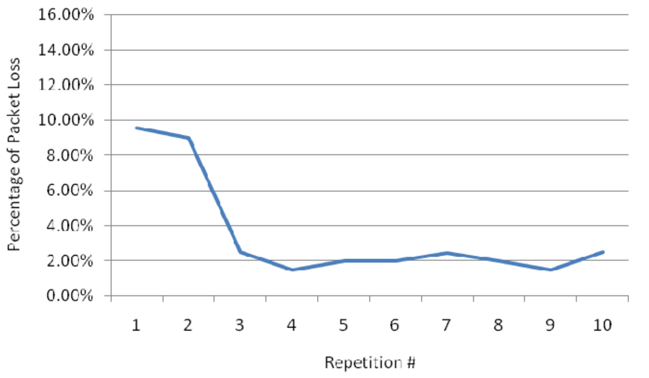

Figure 3. Percentage of packet loss via UDP streaming over standard 802.11g

In our first set of experiments, we measured the number of UDP packets sent on a multicast address from the video streaming server against the number of UDP packets received on the multicast client receiver. Figure 2 and Figure 3 illustrate the results obtained from the ten repetitive experiments that we conducted. As expected, the number of packets received were lesser than the number of packets sent. Since UDP packets are sent without any feedback acknowledgments, this observation is not something new and further validates the fact that UDP transmissions are unreliable. Moreover, the absence of feedback mechanism at the MAC layer due to 802.11's broadcasting treatment upon multicast packets may have contributed to the overall poor streaming quality as well.

Figure 3 shows the percentage of packet loss incurred for the ten repetitive experiments. The average packet loss calculated for all the repetitions was 3.5%. Meanwhile, the standard deviation for the number of packet loss drew up to 3.08%, a relatively higher deviation as compared to the other sets of experiment shown later. It can be inferred that this higher deviation is attributed to UDP being lossy in nature.

B. UDP Streaming over 802.11g with Enhanced Multicast Mechanism

Our second set of experiments was conducted in a similar fashion as the first set of experiments except that this time, the enhanced multicast mechanism feature was enabled on the 802.11 access point. Therefore, each multicast packet identified at the access point would be encapsulated in a frame addressed to the multicast receivers' MAC address. Figure 3 and Figure 4 illustrate the results obtained from said experiment.

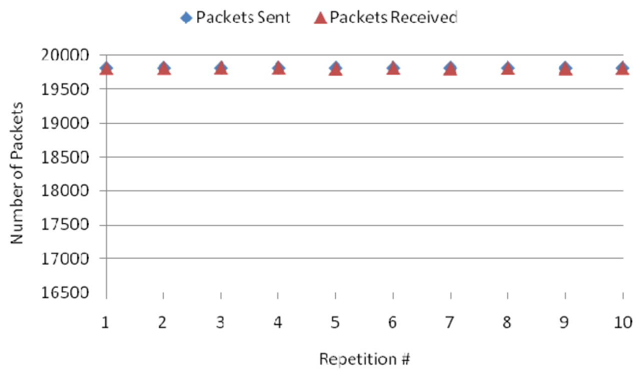

Figure 4. Number of packets sent and received via UDP streaming over 802.11g with enhanced multicast mechanism

The results obtained in Figure 4 shows that despite the usage of UDP protocol, the number of packets sent and received are relatively stable. This proves how significant the effect of the enhanced multicast mechanism is on improving multicast packet transmission.

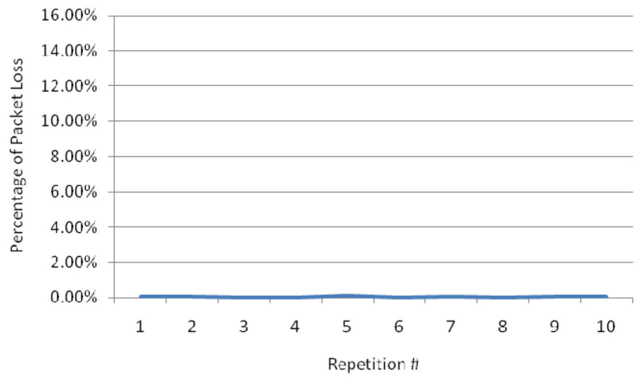

Figure 5. Percentage of packet loss via UDP streaming over 802.11g with enhanced multicast mechanism

Figure 5 shows packet loss of almost 0% for the experiments. The average percentage of packet loss calculated was 0.04% with a standard deviation of 0.03%. This result infers that despite the UDP protocol, multicast traffic throughput can be improved with the multicast enhancement mechanism on the 802.11 protocol at the MAC layer. In addition to this, the result also logically indicates that as long as the lower layer protocol can provide transmission in a seemingly reliable manner, then anything at the upper layers would of course reflect this thereby indicating near-perfect throughput.

C. RTP/UDP Streaming over Standard 802.11g

For our third set of experiments, we used RTP/UDP multicast streaming. At this point, the enhanced multicast mechanism was disabled on the access point. Figure 6 and Figure 7 show the results obtained from this experiment.

Figure 6 displays the number of RTP/UDP packets sent on a multicast address from the video streaming server against the number of RTP/UDP packets received on the multicast client receiver. Compared to streaming on UDP only in the first set of experiments, RTP/UDP streaming seems to show more packet loss. This observation was beyond expectation as we had initially hypothesized that RTP/UDP streaming would be more reliable considering the fact that packet sequencing is part of RTP feature.

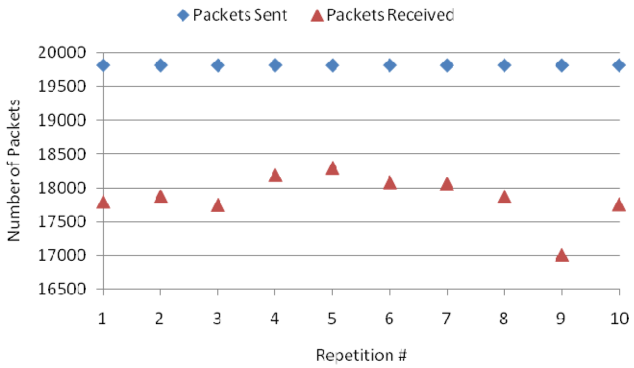

Figure 6. Number of packets sent and received via RTP/UDP streaming over standard 802.11g

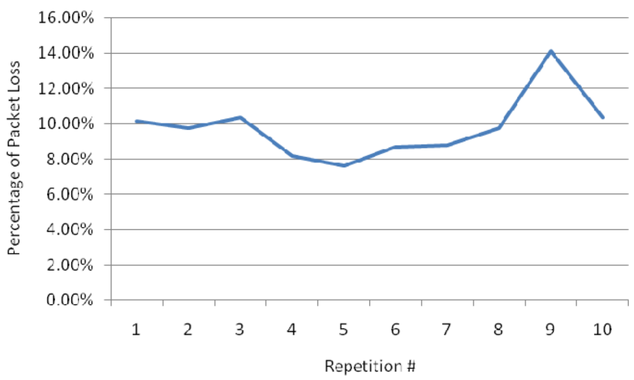

Figure 7. Percentage of packet loss via RTP/UDP streaming over standard 802.11g

Figure 7 shows a relatively fluctuating graph indicating the percentage of packet loss incurred during each of the repetitive experiments. An average packet loss of 9.78% was calculated based on the results obtained with a standards deviation of 1.8%. In essence, the percentage of packet loss is significantly higher than that observed for UDP only streaming during the first set of experiments.

D. RTP/UDP Streaming over 802.11g with Enhanced Multicast Mechanism

The last set of experiments was conducted with the RTP/UDP streaming and the enhanced multicast mechanism enabled on the access point.

Figure 8 shows the number of packets sent and received via RTP/UDP streaming over an 802.11 enhanced multicast mechanism. From our tabulated data, the number of packets received are almost equal to the number of packets sent. Figure 9 shows the percentage of packet loss during this experiment. Average packet loss calculated was 0.02% with standard deviation of 0.02%, an indication that the multicast throughput was more stable in this case. Table I summarizes our overall results.

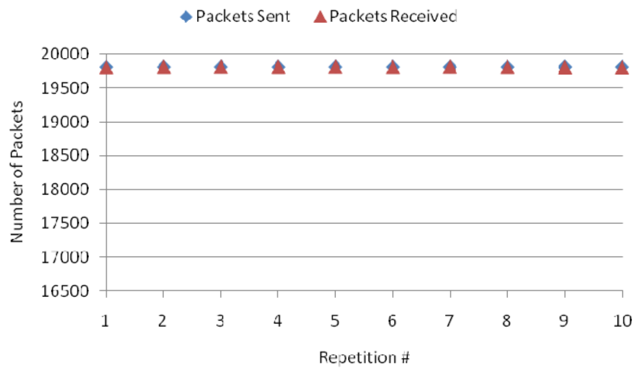

Figure 8. Number of packets sent and received via RTP/UDP streaming over 802.11g with enhanced multicast mechanism

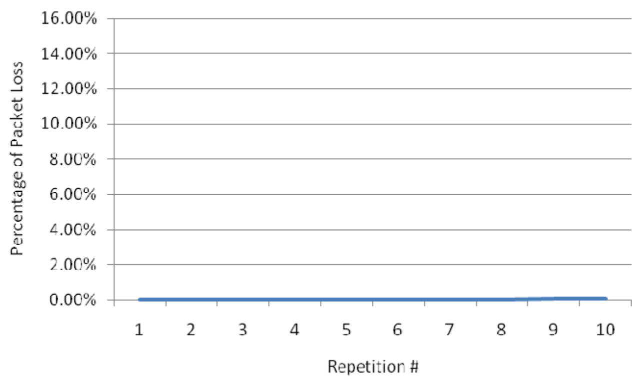

Figure 9. Percentage of packet loss via RTP/UDP streaming over 802.11g with enhanced multicast mechanism

From Table I, it can be deduced that with the 802.11 enhanced multicast mechanism enabled, multicast packet loss can be reduced significantly regardless of whether UDP or RTP/UDP protocol is being used (**Exp. 2** and **Exp. 4**). With the multicast enhancement feature disabled, a significant number of packet loss can be observed when RTP/UDP streaming is being used (**Exp. 3**) as compared to the streaming which uses only UDP (**Exp. 1**). However, when both RTP/UDP streaming and the 802.11 enhanced multicast mechanism are being used (**Exp. 4**), the number of packet loss observed is much lesser than that in the other three experimental setups.

TABLE I. SUMMARY OF AVERAGE PACKET LOSS AND STANDARD DEVIATION PER SET OF EXPERIMENTS

Experiments	Average packet loss	Standard deviation
[**Exp. 1**] UDP streaming only	3.50%	3.08%
[**Exp. 2**] UDP streaming with 802.11 enhanced multicast mechanism	0.04%	0.03%
[**Exp. 3**] RTP/UDP streaming only	9.78%	1.80%
[**Exp. 4**] RTP/UDP with 802.11 enhanced multicast mechanism	0.02%	0.02%

V. CONCLUSION AND FUTURE WORK

We compared UDP and RTP/UDP streaming performance on an 802.11 access point with a MAC layer multicast enhancement mechanism. The 802.11 multicast enhancement feature is configurable and can thus be either enabled or disabled. The UDP and RTP/UDP protocols were tested with and without the 802.11 multicast enhancement feature. For each experimental setup, we captured packets sent from the multicast streaming source and packets received at the multicast receiver, thereby repeating the experiment ten times per setup as the sample size.

The purpose of our study was to prove that with RTP/UDP streaming, multicast throughput can be increased since packet sequence and timestamp fields are available as two of the fields in an RTP header. We used VLC version 0.8.6 to stream and receive multicast packets since from our knowledge and previously gained experience, the application provides support for RTP and is deemed as the most stable amongst other versions available. Further to this study, we also wanted to prove that with the 802.11 multicast enhancement feature enabled, the multicast streaming throughput would improve significantly especially when the RTP protocol is being used.

Our study indicated that with the RTP/UDP being used and the 802.11 multicast enhancement protocol enabled, the multicast streaming throughput prevails as the highest (lesser packet loss) when compared to a similar setup with only the UDP protocol being used and the 802.11 multicast enhancement protocol enabled. However, the difference is only slight and may require a bigger sample size in order to strengthen this finding.

As per the 802.11 multicast enhancement, the feature certainly improves multicast throughput. This is expected as the feature enables transmission of multicast packet via unicast MAC layer headers, which in turn improves the reliability of packets being transmitted. Further extension to this work may consider experimenting on 802.11a, 802.11b and 802.11n as this paper concentrates solely on 802.11g implementation. Apart from that, the number of multicast receivers should be more rather than limiting it to only one. This would reflect a real multicast scenario on the 802.11 standard. Theoretically, a multicast being streamed on an 802.11b/g compliant access point would perform better if the streaming bit rate is below 1Mbps because the base data rate for both 802.11b and 802.11g is 1Mbps. As our future work, we would like to perform experiment on the multicast throughput on 802.11 by varying the streaming bit rate.

VI. REFERENCES

[1] M. Wu, S. Makharia, H. Liu, D. Li, and S. Mathur, “IPTV multicast over wireless LAN using merged hybrid ARQ with staggered adaptive FEC,” IEEE Transactions on Broadcasting, vol. 55, no. 2, pp. 363-374, 2009.

[2] O. Alay, T. Korakis, Y. Wang, E. Erkip, and S.S. Panwar, “Layered wireless video multicast using relays,” IEEE Transactions on Circuits and Systems for Video Technology, vol. 20, no. 8, pp. 1095-1109, 2010.

[3] Q. Du and X. Zhang, “Effective capacity optimization with layered transmission for multicast in wireless networks,” Proceedings of IWCMC ’08. International, 2008, pp. 267-272.

[4] W. Chaohui, L. Yuhong, and M. Jian, “A new reliable multicast scheme for multimedia applications in wireless environment,” 2nd IEEE IC-BNMT 2009., 2009, pp. 744-748.

[5] W. Ge, J. Zhang, and S. Shen, “A cross-layer design approach to multicast in wireless networks,” IEEE Transactions on Wireless Communications, vol. 6, no.3, pp. 1063-1071, March 2007.

[6] Z. Li, T. Herfet, “MAC layer multicast error control for IPTV in wireless LANs,” IEEE Transactions on Broadcasting, vol. 55, no. 2, pp. 353-362, June 2009.

[7] IEEE Std 802.11-2007, Wireless LAN Medium Control (MAC) and Physical Layer (PHY) Specifications.

[8] Y. Tanigawa, K. Yasukawa, and K. Yamaoka, “Transparent Unicast Translation to Improve Quality of Multicast over Wireless LAN,” Proceedings of IEEE CCNC 2010 , January 2010, pp. 1-5.

[9] G. Bianchi, L. Fratta, M. Oliveri, “Performance evaluation and enhancement of the CSMA/CA MAC protocol for 802.11 wireless LANs,”, 7th IEEE PIMRC ’96., vol. 2, 15-18 Oct. 1996, pp. 392-396.

[10] H.S. Chhaya, and S. Gupta, “Performance modeling of asynchronous data transfer methods of IEEE 802.11 MAC protocol,” ACM/Balzer Wireless Networks, vol. 3, pp. 217-234, 1997.

[11] J. Weinmiller, H. Woesner, J.P. Ebert, and A. Wolisz, “Analyzing and Tuning the Distributed Coordination Function in the IEEE 802.11 DFWMAC Draft Standard,” Proceedings of MASCOT ’96, CA, 1996.

[12] J. Weinmiller, M. Schläger, A. Festag, and A. Wolisz, “Performance study of access control in wireless LANs – IEEE 802.11 DFWMAC and ETSI RES 10 HIPERLAN,” Mob. Net, and Appl., v. 2, pp. 55-67, 1997.

[13] F. Boronat, M. Montagud, and V. Vidal, “Enhanced RTP-based Tool-set for Video Streaming Using NS-2,” Proceedings of the 8th MoMM’10, ACM, NY, USA, 2010, pp. 382-386.

[14] J. Postel, “User Datagram Protocol,” RFC 768, Aug. 1980.

[15] A. Sharif, V. Potdar, and A. J. D. Rathnayaka, “Performance Evaluation of Different Transport Layer Protocols on the IEEE 802.11 and IEEE 802.15.4 MAC/PHY Layers for WSN,” Proceedings of the 7th MoMM 2009, ACM, NY, USA, 2009, pp. 300-310.

[16] M. Mathis and J. Mahdavi, “Forward Acknowledgement: Refining TCP Congestion Con-trol,” Proc. of SIGCOMM'96, Martha Steenstrup (Ed.). ACM, NY, USA, 1996, pp. 281-291.

[17] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, “RTP: A Transport Protocol for Real-Time Applications,” RFC 3550, July 2003.

[18] Costeux, J.-L.; Guyard, F.; Bustos, A.-M., "QRP08-5: Detection and Comparison of RTP and Skype Traffic and Performance," IEEE GLOBECOM 2006. IEEE, 2006, pp.1-5.

[19] J. Rey, D. Leon, A. Miyazaki, V. Varsa, and R. Hakenberg, “RTP Retransmission Payload Format,” RFC 4588, Jul. 2006.

[20] http://www.wireshark.org/

[21] MIMOS WiWi Outdoor AP, http://www.mimos.my/2010/07/wiwi-outdoor-access-point-coming-soon/